MODULE 42
NETWORK PROGRAMMING
SOCKET PART IV
Advanced TCP/IP and RAW SOCKET

My Training Period: hours

Note:
This is a continuation from Part III, Module41. Working program examples compiled using `gcc`, tested using the public IPs, run on Fedora 3, with several times of update, as root or suid 0. The Fedora machine used for the testing having the "**No Stack Execute**" disabled and the **SELinux** set to default configuration.
This Module will concentrate on the TCP/IP stack and will try to dig deeper till the packet level.

Abilities

- Able to understand the 7 layers OSI stack.
- Able to understand the 4 layers TCP/IP stack/suite/layer.
- Able to understand protocols in TCP/IP stack.
- Able to find and appreciate the RFCs and Standards.
- Able to understand and use the RAW socket (vs cooked socket).
- Able to understand and use for good purposes of the useful network tools that can be developed using RAW socket.

Introduction

In the previous Modules we just dealt with the generic TCP/UDP programs. It is not so usable without implementing other protocols at other layers of the TCP/IP suite. In this Module we will investigate deeper into the TCP/IP suite, their protocols, header formats and at the end we will try to construct our own packet using RAW packet. Let recall some of the information that we have already covered in the previous Modules and then proceed on to the details. The following figure shows various TCP/IP and other protocols reside in the original OSI model.

7	**Application**	e.g. HTTP, SMTP, SNMP, FTP, Telnet, SSH and Scp, NFS, RTSP etc.
6	**Presentation**	e.g. XDR, ASN.1, SMB, AFP etc.
5	**Session**	e.g. TLS, SSH, ISO 8327 / CCITT X.225, RPC, NetBIOS, ASP etc.
4	**Transport**	e.g. TCP, UDP, RTP, SCTP, SPX, ATP etc.
3	**Network**	e.g. IP/IPv6, ICMP, IGMP, X.25, CLNP, ARP, RARP, BGP, OSPF, RIP, IPX, DDP etc.
2	**Data Link**	e.g. Ethernet, Token ring, PPP, HDLC, Frame relay, ISDN, ATM, 802.11 Wi-Fi, FDDI etc.
1	**Physical**	e.g. wire, radio, fiber optic etc.

Figure 1: OSI layer.

As discussed in the previous Module, in implementation, the de facto standard used is the TCP/IP. This TCP/IP term should be general and here we will study the detail of the TCP/IP.

TCP/IP

The TCP/IP suite attempts to create a heterogeneous network with open protocols that are independent of operating system and architectural difference. TCP/IP protocols are available to everyone, and are developed and changed by consensus, not by one manufacturer. Everyone is free to develop products to meet these open protocol specifications. Most information about TCP/IP is published as **Request For Comments (**RFC**)**, which contain the latest version of the specifications of all TCP/IP protocols standard. The following figure shows the 4 layers of TCP/IP suite.

4	**Application layer**	BGP, FTP, HTTP, HTTPS, IMAP, IRC, NNTP, POP3, RTP, SIP, SMTP, SNMP, SSH, SSL, Telnet, UUCP, Finger, Gopher, DNS, RIP, Traceroute, Whois, IMAP/IMAP4, Ping, RADIUS, BGP etc.
3	**Transport layer.**	DCCP, OSPF, SCTP, TCP, UDP, ICMP etc.
2	**Network/Internet layer.**	IPv4, IPv6, ICMP, ARP, IGMP etc
1	**Physical/ Data Link layer.**	Ethernet, Wireless (WAP, CDPD, 802.11, Wi-Fi), Token ring, FDDI, PPP, ISDN, Frame Relay, ATM, SONET/SDH,

	xDSL, SLIP etc. RS-232, EIA-422, RS-449, EIA-485 etc.

Figure 2: TCP/IP stack/layer/suite.

Commonly, the top three layers of the OSI model (Application, Presentation and Session) are considered as a single Application layer in the TCP/IP suite and the bottom two layers as well considered as a single Network Access layer. Because the TCP/IP suite has no unified session layer on which higher layers are built, these functions are typically carried out (or ignored) by individual applications. The most notable difference between TCP/IP and OSI models is the Application layer, as TCP/IP integrates a few steps of the OSI model into its Application layer. A simplified TCP/IP interpretation of the stack is shown below:

Application	e.g. HTTP, FTP, DNS. (Routing protocols like BGP and RIP, which for a variety of reasons run over TCP and UDP respectively, may also be considered part of the Network layer)
Transport	e.g. TCP, UDP, RTP, SCTP. (Routing protocols like OSPF, which run over IP, may also be considered part of the Network layer)
Network/Internet	For TCP/IP this is the Internet Protocol (IP). (Required protocols like ICMP and IGMP run over IP, but may still be considered part of the network layer; ARP does not run over IP).
Physical/ Data Link/ Network Access	e.g. Ethernet, Token ring, etc. e.g. physical media, and encoding techniques, T1, E1 etc.

Figure 3: Brief of the TCP/IP stack functions.

The basic function each of the TCP/IP layer is illustrated in the following figure.

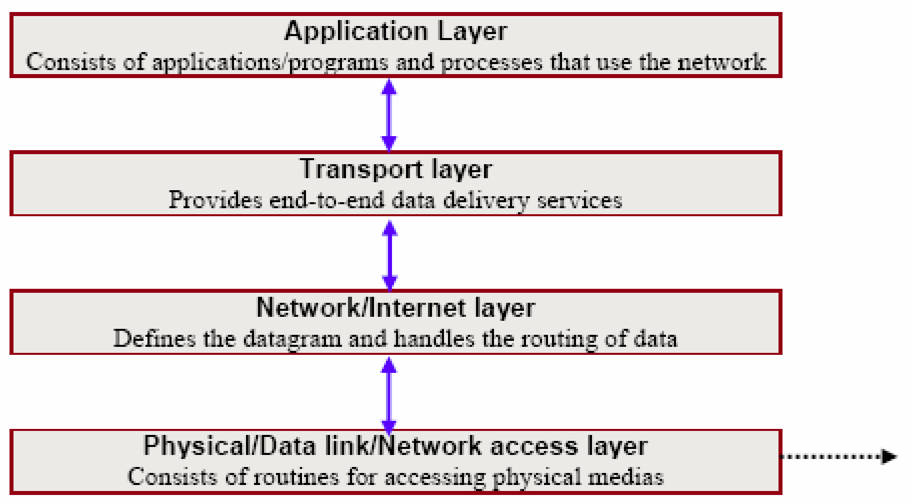

Application Layer
Consists of applications/programs and processes that use the network
Transport layer
Provides end-to-end data delivery services
Network/Internet layer
Defines the datagram and handles the routing of data
Physical/Data link/Network access layer
Consists of routines for accessing physical medias

Figure 4: Another TCP/IP basic stack functionalities.

A shown in figure 5, the four-layered structure of TCP/IP is seen in the way data handled as it passes down the protocol stack from the Application layer to the underlying physical network. Each layer in the stack adds control information to ensure proper delivery. This control information is called a **header** because it is placed in front of the data to be transmitted. Each layer treats all of the information it receives from the layer above as **data** and places its own header in front of that information. The addition of delivery information at every layer is called **encapsulation**. Note that the **real data** that will be transmitted, seen or used at Application layer just a small portion of the whole packet. When data is received, the opposite process happens. Each layer strips off its header before passing the real data on the layer above. As information flows back up the stack, information received from a lower layer is interpreted as both a header and data.

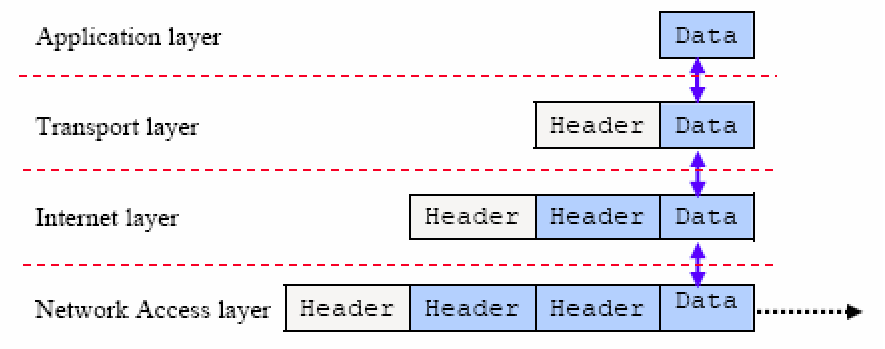


Figure 5: TCP/IP header encapsulation.

Each layer has its own independent data structures. Conceptually a layer is unaware of the data structure used by the layers above and below it. In reality, the data structures of a layer are designed to be compatible with the structures used by the surrounding layers for the sake of more efficient data transmission. Still, each layer has its own data structure and its own terminology to describe that structure. Figure 6 shows the terms used by different layers of TCP/IP to refer to the data being transmitted. As a general term, most networks refer to a transmitted data as **packets** of **frames**.

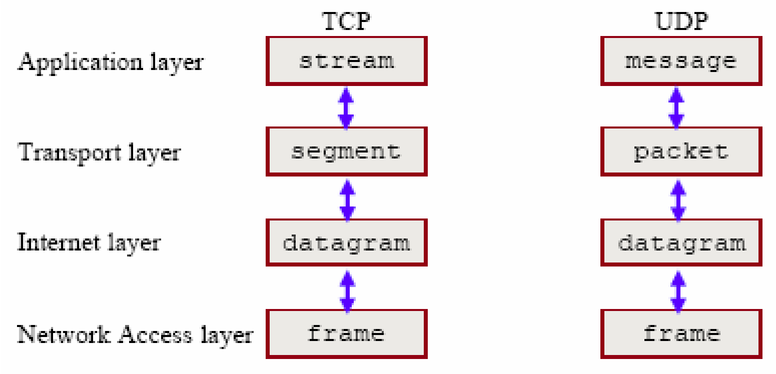


Figure 6: Different term of packet at different TCP/IP layers.

TCP/IP: The Detail

The following figure tries to give a big picture of what actually happen when a host (network device) communicates with another host in TCP/IP stack. The flow of the packets is two ways representing the terms send and receive. Using figure 7 as our reference, let investigate more detail of every layer starting from the bottom layer of the TCP/IP stack.

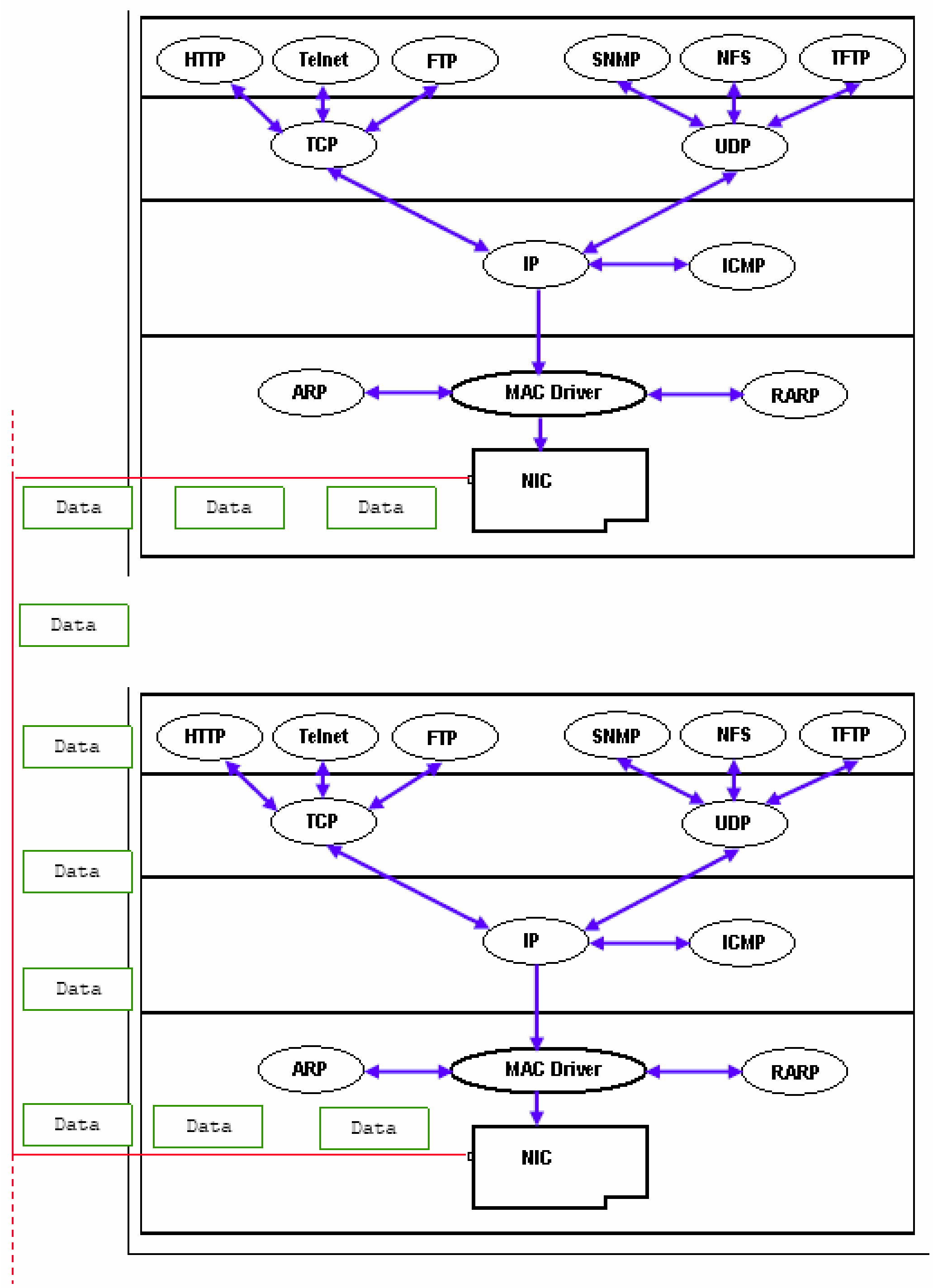


Figure 7: Quite a complete structure of protocols diagram used for communication.

Network Access Layer

The Network Access layer it is the lowest layer of the TCP/IP protocol hierarchy. The protocols in this layer provide the means for the system to deliver data to the other device on a directly attached network. It defines how to use the network to transmit an IP diagram. Unlike higher-level protocols, it must know the details of the underlying network to correctly format the data being transmitted to comply with the network constraints. The TCP/IP Network Access layer can encompass the function of all three lower layers of the OSI reference model Network layer, Data Link layer, and Physical layer.

Functions performed at this level include encapsulation of IP datagrams into the frames transmitted by the network and mapping of IP addresses to the physical addresses used by the network (provided by ARP protocol). The network access layer is responsible for exchanging data between a host and the network and for delivering data between two devices on the same network. Node physical addresses (MAC address) are used to accomplish delivery on the local network.

TCP/IP has been adapted to a wide variety of network types, including switching, such as X.21, packet switching, such as X.25, Ethernet, the IEEE 802.x protocols, frame relay, wireless etc. For example, data in the network access layer encode `EtherType` (Ethernet) information that is used to demultiplex data associated with specific upper-layer protocol stacks.

Network/Internet Layer

The Internet layer is the heart of TCP/IP and the most important protocol. This layer provides the basic packet delivery service on which TCP/IP networks are built. The TCP/IP protocol at this layer is the **Internet Protocol** (IP- RFC 791). **All** protocols, in the layers above and below Internet layer, use the **Internet Protocol** to deliver data. **All** TCP/IP data flows through IP, incoming and outgoing, regardless of its final destination.
The Internet layer is responsible for **routing** messages through **internetworks**. Devices responsible for routing messages between networks are called **gateways** in TCP/IP terminology, although the term **router** is also used with increasing frequency. In addition to the physical node addresses utilized at the network access layer, the IP protocol implements a system of **logical host** addresses called **IP addresses**. The IP addresses are used by the **internet** and **higher layers** to identify devices and to perform internetwork routing. As discussed in the previous Module the IP address may be a class or classless type. The **Address Resolution Protocol (ARP)** enables IP to identify the physical address (Media Access Control, MAC) that matches a given IP address. The physical address has been burnt on every NIC. To make it readable for human being, the (domain) name is used instead of the IP address in normal operation. The IP address and name resolution is done by Domain Name System (DNS). In the implementation, UNIX/Linux uses BIND and Windows uses Domain Name Service (also DNS acronym). The relationship is shown in the following figure.

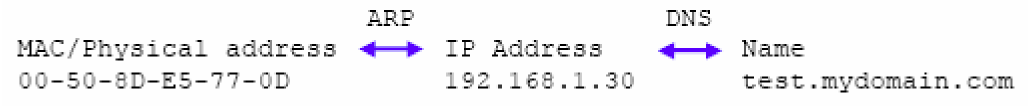


Figure 8: IP/Name resolution.

The IP provides services that are roughly equivalent to the OSI Network layer. IP provides a datagram (connectionless) transport service across the network. This service is sometimes referred to as **unreliable** because the network does not guarantee delivery nor notify the end host system about packets lost due to errors or network congestion. IP datagrams contain a message, or one fragment of a message, that may be up to 65,535 bytes (octets/bytes) in length. IP does not provide a mechanism for flow control. Let dig deeper about the protocols in this layer.

Internet Protocol (IP)

The IP protocol functionalities include:

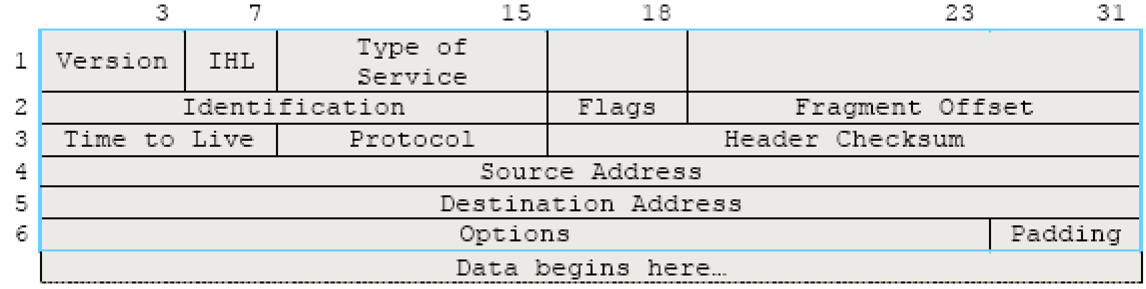
- Defining the datagram, which is the basic unit of transmission in the Internet.
- Defining the Internet addressing scheme, moving data between the Network Access layer and the Transport layer.
- Routing datagrams to remote hosts.
- Performing fragmentation and reassembly of datagrams.

The Datagram

Is a packet format defined by Internet Protocol. The internet protocol delivers the datagram by checking the **Destination Address (DA)**. This is an IP address that identifies the destination network and the specific host on that network. If the destination address is the address of a host on the local network, the packet is delivered directly to the destination; otherwise the packet is passed to a gateway/router for delivery. Gateways are devices that switch packets between the different physical networks. Deciding which gateway to use is called **routing**. IP makes the routing decision for each individual packet. IP deals with data in chunks called **datagrams**. The terms packet and datagram are often used interchangeably, although a packet is a data link-layer object and a datagram is a network layer object. In many cases, particularly when using IP on Ethernet, a datagram and packet refer to the same chunk of data. There's no guarantee that the physical link layer can handle a packet of the network layer's size. If the media's MTU is smaller than the network's packet size, then the network layer has to break large datagrams down into packed-sized chunks that the data link layer and physical layer can digest. This process is called **fragmentation**. The host receiving a fragmented datagram reassembles the pieces in the correct order.

IPv4 Datagram Format

The following figure shows the IPv4 datagram header format. It is 6 x 32 bits (word size) wide.



	3	7	15	18	23 31
1	Version	IHL	Type of Service		
2	Identification			Flags	Fragment Offset
3	Time to Live		Protocol	Header Checksum	
4	Source Address				
5	Destination Address				
6	Options				Padding
	Data begins here…				

Figure 9: The IP Datagram Format.

A brief field description:

Field	Description
Version	The version of IP currently used.
IHL	IP Header Length (IHL) - datagram header length. Points to the beginning of the data. The minimum value for a correct header is 5.
Type of Service	Data in this field indicate the quality of service desired. The effects of values in the precedence fields depend on the network technology employed, and values must be configured accordingly. Format of the Type of Service field: ▪ Bits 0-2: Precedence 111 = Normal Control. 110 = Internetwork Control. 101 = CRITIC/ECP. 100 = Flash Override. 011 = Flash. 010 = Immediate. 001 = Priority. 000 = Routine. ▪ Bit 3 : Delay 0 = normal delay, 1 = low delay. ▪ Bit 4 : Throughput 0 = normal throughput, 1 = high throughput. ▪ Bit 5 : Reliability 0 = normal reliability, 1 = high reliability. ▪ Bits 6-7: Reserved
Total Length	The length of the datagram in byte, including the IP header and data. This field enables datagrams to consist of up to 65,535 bytes. The standard recommends that all hosts be prepared to receive datagrams of at least 576 bytes in length.
Identification	An identification field used to aid reassembles of the fragments of a datagram.
Flags	If a datagram is fragmented, the MB bit is 1 in all fragments except the last. This field contains three control bits: ▪ Bit 0: Reserved, must be 0. ▪ Bit 1 (**DF**): 1 = **D**o not **f**ragment and 0 = May fragment. ▪ Bit 2 (**MF**): 1 = **M**ore **f**ragments and 0 = Last fragment.
Fragment Offset	For fragmented datagrams, indicates the position in the datagram of this fragment.
Time-to-live	Indicates the maximum time the datagram may remain on the network.
Protocol	The 8 bits field of the upper layer protocol associated with the data portion of the datagram. For a complete information please refer to RFC 1700 and the following is some of the protocol numbers: **Decimal** **Protocol** 1 ICMP (Internet Control Message) 2 IGMP (Internet Group Management) 4 IP (IP in IP -encapsulation) 5 ST (Stream) 6 TCP (Transmission Control) 17 UDP (User Datagram) 27 RDP (Reliable Data Protocol)
Header Checksum	A checksum for the header only. This value must be recalculated each time the header is

	modified.
Source Address	The IP address of the originated the datagram.
Destination Address	The IP address of the host that is the final destination of the datagram.
Options	May contain 0 or more options.
Padding	Filled with bits to ensure that the size of the header is a 32-bit multiple.

Table 1: IP datagram fields description.

Note that in the IP packet we just have the source and destination IP addresses. There is no source and destination port numbers here which is set in UDP or TCP header.

Internet Control Message Protocol (ICMP and ICMPv6)

Is part of the **Internet layer** and **uses the IP datagram delivery facility** to sends its messages. ICMP sends messages that perform control, error reporting, and informational functions for TCP/IP. The RFC document for ICMP is RFC 792. The following figure is the ICMP header format.

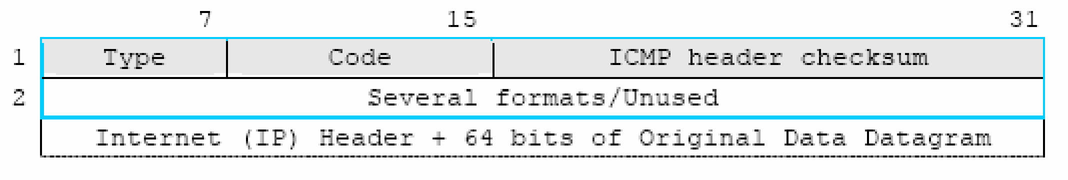


Figure 10: The ICMP Header Format.

A brief description:

Field	Description
Type	Messages can be error or informational messages. Error messages can be Destination unreachable, Packet too big, Time exceed, Parameter problem. The possible informational messages are, Echo Request, Echo Reply, Group Membership Query, Group Membership Report and Group Membership Reduction. A summary of message Types are listed below. 0: Echo Reply. 3: Destination Unreachable. 4: Source Quench. 5: Redirect. 8: Echo. 11: Time Exceeded. 12: Parameter Problem. 13: Timestamp. 14: Timestamp Reply. 15: Information Request. 16: Information Reply.
Code	For each type of message as listed above, several different codes are defined. An example of this is the Destination Unreachable message, where possible messages are: no route to destination, communication with destination administratively prohibited, not a neighbor, address unreachable, port unreachable. The code and its means for Destination Unreachable message is listed below. 0 = net unreachable. 1 = host unreachable. 2 = protocol unreachable. 3 = port unreachable. 4 = fragmentation needed and DF set. 5 = source route failed.
Checksum	The 16-bit one's complement of the one's complement sum of the ICMP message starting with the ICMP Type. For computing the checksum, the checksum field should be zero.
Second word (Several	Several formats that match with certain IP header fields/depend on the Type and Code fields.

formats/unused	

Table 2: ICMP datagram fields description.

The usage examples of the ICMP (together with IP) are listed below:

- Flow control: When datagrams arrive too fast for processing, the destination host or intermediate gateway sends an `ICMP Source Quench Message` back to the sender. This tells the source to temporarily stop sending datagrams.
- Detecting unreachable destinations: When a destination is unreachable, the system detecting the problem sends an `ICMP Destination Unreachable Message` to the datagrams source. If the unreachable destination is a network or host, the message is sent by an intermediate gateway. But if the destination is an unreachable port, the destination host sends the message.
- Redirecting routes: A gateway sends the `ICMP Redirect Message` to tell a host to use another gateway, presumably because the other gateway is a better choice. This message can only be used when the source host is on the same network as both gateways.
- Checking remote hosts: A host can send the ICMP `Echo Message` to see if a remote system's internet protocol is up and operational. When a system receives an echo message, it sends the same packet back to the source host (e.g. **PING** command).

Other message types include:

- Information Request or Information Reply Message.
- Timestamp or Timestamp Reply Message.
- Parameter Problem Message.
- Time Exceeded Message.

Unless otherwise noted under the individual format descriptions as explained above, the values of the Internet Protocol (IP) header fields for the ICMP are as follows:

IP Field	Description
`Version`	4.
`IHL`	Internet header length in 32-bit words.
`Type of Service`	0.
`Total Length`	Length of internet header and data in octets.
`Identification, Flags, Fragment Offset`	Used in fragmentation.
`Time to Live`	Time to live in seconds; as this field is decremented at each machine in which the datagram is processed, the value in this field should be at least as great as the number of gateways which this datagram will traverse.
`Protocol`	ICMP = 1.
`Header Checksum`	The 16 bit one's complement of the one's complement sum of all 16 bit words in the header. For computing the checksum, the checksum field should be zero. This checksum may be replaced in the future.
`Source Address`	The address of the gateway or host that composes the ICMP message. Unless otherwise noted, this can be any of a gateway's addresses.
`Destination Address`	The address of the gateway or host to which the message should be sent.

Table 3: IP fields description when used with ICMP.

(Host-to-Host) Transport Layer

The Transport layer has two major jobs:

1. It must subdivide user-sized data buffers into network layer sized datagrams, and
2. It must enforce any desired transmission control such as reliable delivery.

The Transport layer is responsible for end-to-end data integrity. The two most important protocols in this layer are **Transmission Control Protocol (TCP)** and **User Datagram Protocol (UDP)**. TCP provides reliable data delivery service with end-to-end error detection and correction and also enables hosts to maintain multiple, simultaneous connections. UDP provides low-overhead, connectionless datagram delivery service. Both protocols deliver data between the Application layer and the Internet layer. Applications programmers can choose whichever service is more appropriate for their specific applications. Protocols defined at this layer accept data from application protocols running at the Application layer, encapsulate it in the protocol header, and deliver the data segment thus formed to the lower IP layer for routing. Unlike the IP protocol, the transport layer is aware of the identity of the ultimate user representative process. As such, the Transport layer, in the TCP/IP suite, embodies what data communications are all about: The delivering of information from an application on one computer to an application on another computer.

User Datagram Protocol (UDP – RFC768)

Gives application programs direct access to a datagram delivery service, like the delivery service that IP provides. This allows applications to exchange messages over the network with a minimum of protocol overhead. UDP is an unreliable (it doesn't care about the quality if deliveries it make), connectionless (doesn't establish a connection on behalf of user applications) datagram protocol. Within your computer, UDP will deliver data correctly. UDP is used as a data transport service when the amount of data being transmitted is small, the overhead of creating connections and ensuring reliable delivery may be greater than the work of retransmitting the entire data set. Broadcast-oriented services use UDP, as do those in which repeated, out of sequence, or missed requests have no harmful side effects. Since no state is maintained for UDP transmission, it is ideal for repeated, short operations such as the **Remote Procedure Call** (RPC) protocol. UDP packets can arrive in any order. If there is a network bottleneck that drops packets, UDP packets may not arrive at all. It's up to the application built on UDP to determine that a packet was lost, and to resend it if necessary.

NFS and **NIS** are built on top of UDP because of its speed and statelessness. While the performance advantages of a fast protocol are obvious, the stateless nature of UDP is equally important. Without state information in either the client or server, crash recovery is greatly simplified.

UDP is also the transport protocol for several well-known application-layer protocols, including **Network File System** (NFS), **Simple Network Management Protocol** (SNMP), **Domain Name System** (DNS), and **Trivial File Transfer Protocol** (TFTP). The following figure shows the UDP datagram format.

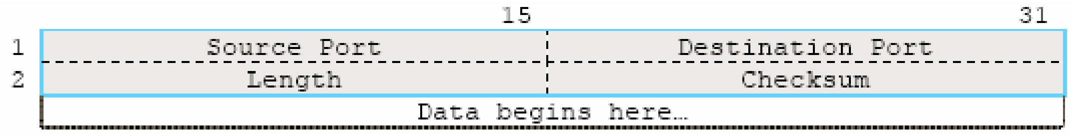


Figure 11: The UDP Datagram Format.

A brief description:

Field	Description
Source Port (16 bits)	This field is optional and specifies the port number of the application that is originating the user data.
Destination Port (16 bits)	This is the port number pertaining to the destination application.
Length (16 bits)	This field describes the total length of the UDP datagram, including both data and header information.
UDP checksum (16 bits)	Integrity checking is optional under UDP. If turned on, this field is used by both ends of the communication channel for data integrity checks.

Table 4: UDP fields description.

Well, let revised what we have already covered till now. The following figure is the TCP/IP stack mentioned before. When data is sent from a host to another host, depend on the application (protocols), it has to go through the layers. Every layer will encapsulate the appropriate header.

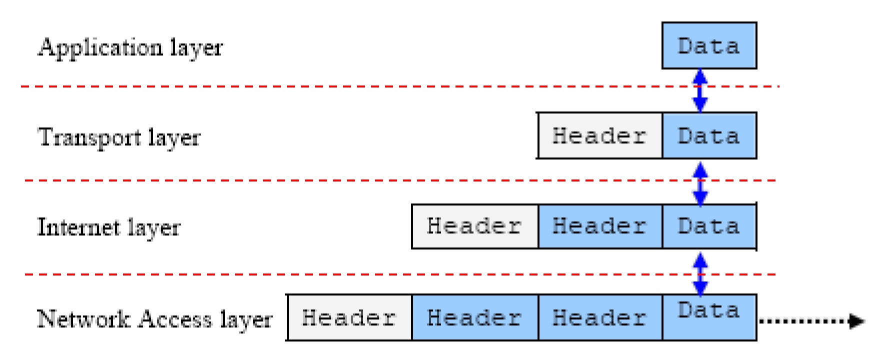


Figure 12: TCP/IP header encapsulation, illustrated vertically.

To make it clearer, the following figure is a packet that horizontally rearranged of the previous figure. The `Data...` may also contain other upper protocol header, the Transport layer.

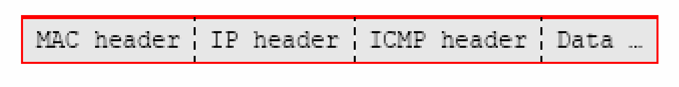
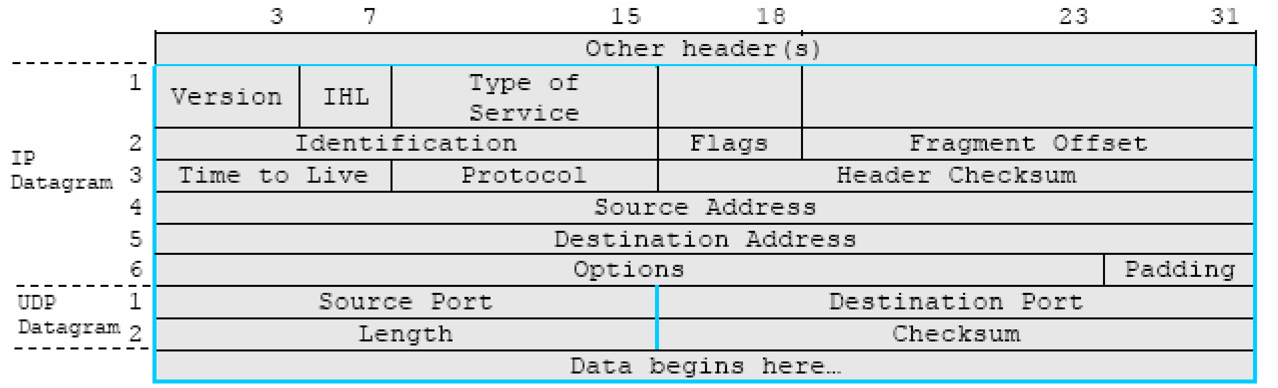


Figure 13: TCP/IP header encapsulation, illustrated horizontally.

As an example, by assuming there is no other information inserted between the Transport and the Internetwork layers, the following figure shows the packet when the data has gone through the Transport and Internerwork layers.



		3	7	15	18	23	31
		Other header(s)					
IP Datagram	1	Version	IHL	Type of Service			
	2	Identification			Flags	Fragment Offset	
	3	Time to Live	Protocol		Header Checksum		
	4	Source Address					
	5	Destination Address					
	6	Options					Padding
UDP Datagram	1	Source Port			Destination Port		
	2	Length			Checksum		
		Data begins here…					

Figure 14: The UDP and IP headers in a packet.

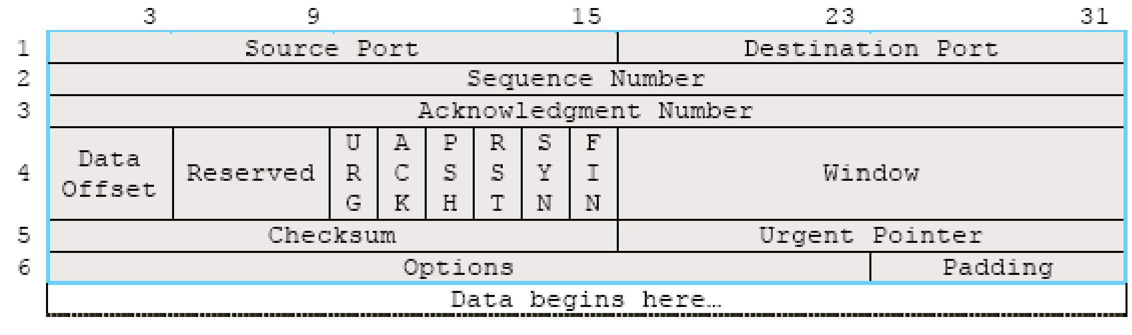
From the above figure, what IP considers to be data field is in fact just another piece of formatted information including both UDP header and user protocol data. To IP it should not matter what the data field is hiding. The details of the header information for each protocol should clearly convey to the reader purpose of the protocol. Keep in mind that at machine level, all the fields in the packet actually just a combination of the 0s and 1s digits. Let continue with another important protocol in Transport layer, the TCP.

Transmission Control Protocol (TCP)

Transmission Control Protocol (TCP) is a required TCP/IP standard defined in RFC 793, "Transmission Control Protocol (TCP)", that provides a reliable, connection-oriented packet delivery service. The Transmission Control Protocol:

- Guarantees delivery of IP datagrams.
- Performs segmentation and reassembly of large blocks of data sent by programs.
- Ensures proper sequencing and ordered delivery of segmented data.
- Performs checks on the integrity of transmitted data by using checksum calculations.
- Sends positive messages depending on whether data was received successfully. By using selective acknowledgments, negative acknowledgments for data not received are also sent.
- Offers a preferred method of transport for programs that must use reliable session-based data transmission, such as client/server database and e-mail programs.

It is fully reliable, connection-oriented, end-to-end packet delivery, acknowledged, byte stream protocol that provide consistency for data delivery across the network in a proper sequence. TCP supports data fragmentation and reassemble. It also support multiplexing/demultiplexing using source and destination port numbers in much the same way they are used by UDP. Together with the Internet Protocol (IP), TCP represents the heart of the Internet protocols. TCP provides reliability with a mechanism called **Positive Acknowledgement with Retransmission (PAR)**. Simply said, a system using PAR resends the data, unless it hears from the remote system that the data received is okay. The unit of data exchanged between co-operating TCP modules is called a **segment**. The following is a TCP segment format.



	3	9						15	23	31
1	Source Port								Destination Port	
2	Sequence Number									
3	Acknowledgment Number									
4	Data Offset	Reserved	URG	ACK	PSH	RST	SYN	FIN	Window	
5	Checksum								Urgent Pointer	
6	Options									Padding
	Data begins here…									

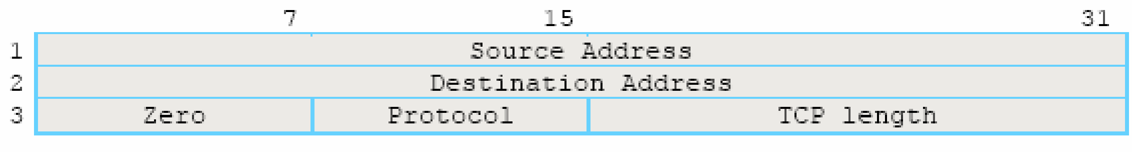
Figure 15: The segment format of the TCP Protocol.

A brief field description:

Field	Description
Source port (16 bits)	Specifies the port on the sending TCP module.
Destination port (16 bits)	Specifies the port on the receiving TCP module.
Sequence number (32 bits)	Specifies the sequence position of the first data octet in the segment. When the segment opens a connection, the sequence number is the **Initial Sequence Number (ISN)** and the first octet in the data field is at sequence ISN+1.
Acknowledgement number 32 bits)	Specifies the next sequence number that is expected by the sender of the segment. TCP indicates that this field is active by setting the ACK bit, which is always set after a connection is established.
Data offset 4 bits)	Specifies the number of 32-bit word in the TCP header.
Reserved (6 bits)	Must be zero. Reserved for future use.
Control bits (6 bits)	The six control bits are as follow: ▪ URG - When set, the Urgent Pointer field is significant. ▪ ACK - When set, the acknowledgement Number field is significant. ▪ PSH - Initiates a push function. ▪ RST - Forces a reset of the connection. ▪ SYN - Synchronizes sequencing counters for the connection. This bit is set when a segment request opening of a connection. ▪ FIN - No more data. Closes the connection.
Window (16 bits)	Specifies the number of octets, starting with the octet specified in the acknowledgement number field, which the sender of the segment can currently accept.
Checksum (16 bits)	An error control checksum that covers the header and data fields. It does not cover any padding required to have the segment consists of an even number of octets. The checksum also covers a 96-pseudoheader as shown below; it includes source and destination addresses, the protocol, and the segment length. The information is forwarded with the segment to IP to protect TCP from miss-routed segments. The value of the segment length a field includes the TCP header and data, but doesn't include the length of the pseudoheader.
Urgent Pointer (16 bits)	Identifies the sequence number of the octet following urgent data. The urgent pointer is a positive offset from the sequence number of the segment.
Options	Options are available for a variety of functions.

(variable)	
Padding (variable)	0-value octets are appended to the header to ensure that the header ends on a 32-bit word boundary.

Table 5: TCP segment fields description.



	7	15	31
1	Source Address		
2	Destination Address		
3	Zero	Protocol	TCP length

Figure 16: The format of the TCP pseudoheader.

TCP Three-Way Handshake

Each segment contains a checksum that the recipient uses to verify that the data is undamaged. If the data segment is received undamaged, the receiver sends a positive acknowledgement back to the sender. If the data segment is damaged, the receiver discards it. After an appropriate time-out period, the sending TCP module retransmits any segment for which no positive acknowledgement has been received. TCP is connection-oriented. It establishes a logical end-to-end connection between the two communication hosts. Control information, called a **handshake**, is exchanged between the two endpoints to establish a dialogue before data is transmitted. TCP indicates the control function of a segment by setting the appropriate bit in the `flags` field of the segment header. The type of handshake used by TCP is called a **three-way handshake** because three segments are exchanged. The following figure illustrates the three-way handshake mechanism.

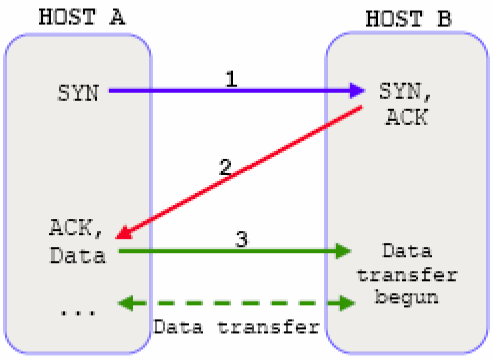


Figure 17: A Three-Way Handshake of the TCP initialization.

The client who needs to initialize a connection sends out a SYN segment (Synchronize) to the server along with the initial sequence number. No data is sent during this process, and the SYN segment contains only TCP header and IP header. When the server receives the SYN segment, it acknowledges the request with its own SYN segment, called SYN-ACK segment. When the client receives the SYN-ACK, it sends an ACK for the server's SYN. At this stage the connection is "established."

Unlike TCP connection initialization, which is a three-way process, connection termination takes place with the exchange of four-way packets. The following figure illustrates the TCP termination process.

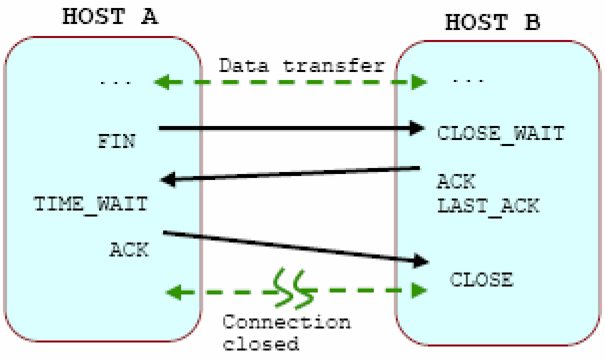


Figure 18: A Four-Way of the TCP termination.

1. The client who needs to terminate the connection sends a `FIN` segment to the server that is a TCP Packet with the `FIN` flag set, indicating that it has finished sending the data.
2. The server, upon receiving the `FIN` segment, does not terminate the connection but enters into a "passive close" (`CLOSE_WAIT`) state and sends an `ACK` for the `FIN` back to the client with the sequence number incremented by one. Now the server enters into `LAST_ACK` state.
3. When the client gets the last `ACK` from the server, it enters into a `TIME_WAIT` state, and sends an `ACK` back to the server with the sequence number incremented by one.
4. When the server gets the `ACK` from the client, it closes the connection.

Reliability and Acknowledgement

TCP employs the positive acknowledgement with retransmission technique for the purpose of achieving reliability in service.

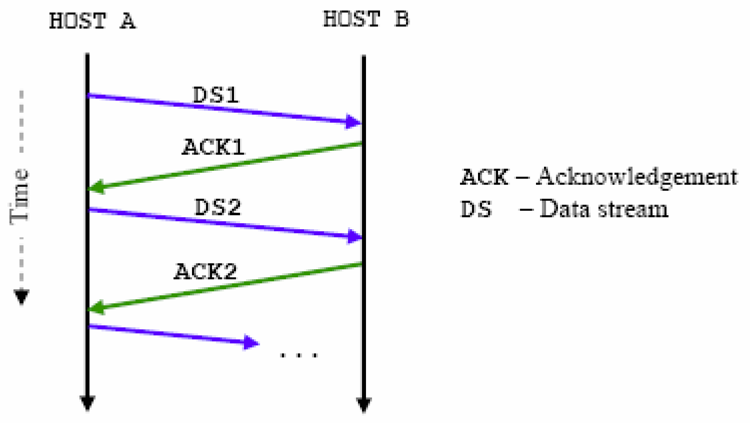


Figure 19: The positive acknowledgement with retransmission technique.

Figure 80 illustrates a simple ladder diagram depicting the events taking place between two hosts. The arrows represent transmitted data and/or acknowledgements, and time is represented by the vertical distance down the ladder. When TCP send a data segment, it requires an acknowledgement from the receiving end. The acknowledgement is used to update the connection state table. An acknowledgement can be positive or negative. A positive acknowledgement implies that the receiving host recovered the data and that it passed the integrity check. A negative acknowledgement implies that the failed data segment needs to be retransmitted. It can be caused by failures such as data corruption or loss.

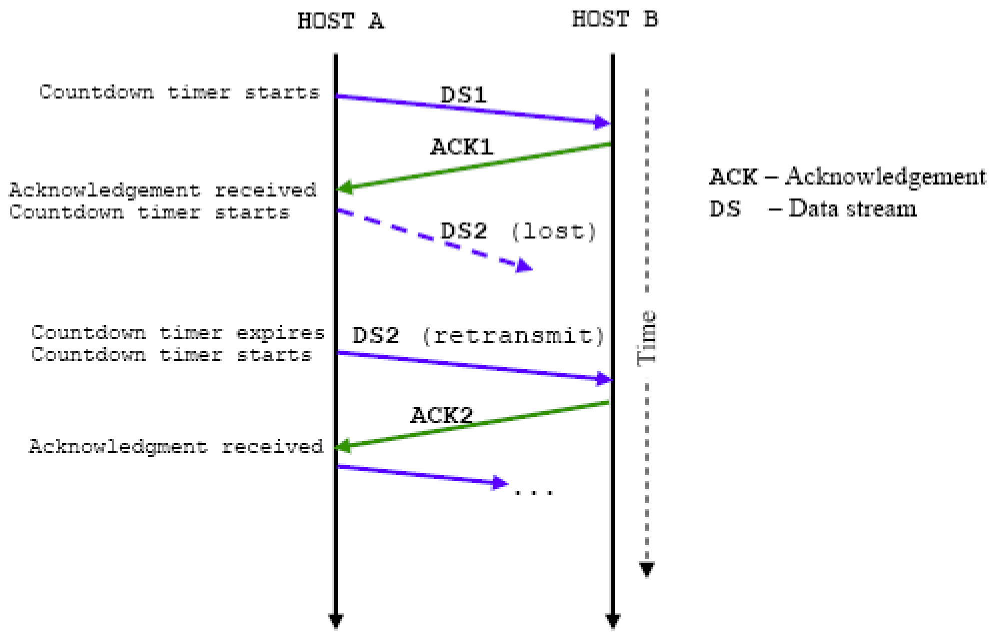


Figure 20: TCP implementation of the time-out mechanism to keep track of loss segments.

In figure 81, illustrates what happens when a packet is lost on the network and fails to reach its ultimate destination. When a host sends data, it starts a countdown timer. If the timer expires without receiving an acknowledgement, this host assumes that the data segment was lost. Consequently, this host retransmits a duplicate of the failing segment. TCP keep a copy of all transmitted data with outstanding positive acknowledgement. Only after receiving the positive acknowledgement is this copy discarded to make room for other data in its buffer.

Data Stream Maintenance

The interface between TCP and a local process is a port, which is a mechanism that enables the process to call TCP and in turn enables TCP to deliver data streams to the appropriate process. Ports are identified by port numbers. To fully specify a connection, the host IP address is appended to the port number. This combination of IP address and port number is called a **socket**. A given socket number is unique on the internetwork. A connection between two hosts is fully described by the sockets assigned to each end of the connection.

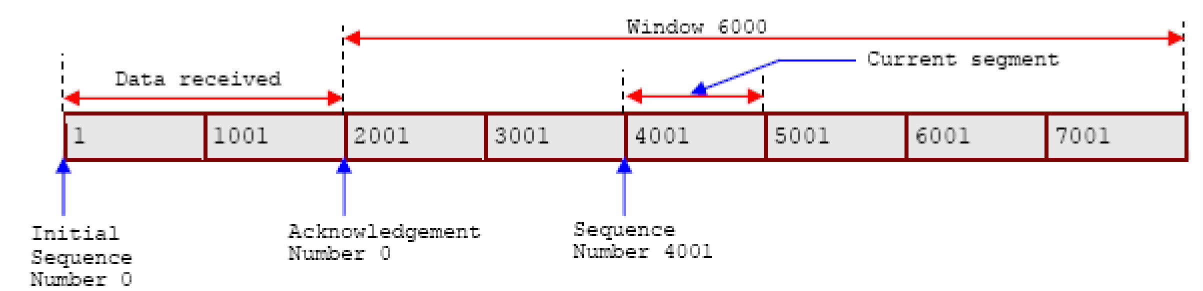


Figure 21: A TCP data stream that starts with an Initial Sequence Number (ISN) of 0.

In figure 82, the receiving system has received and acknowledged 2000 bytes. So the current Acknowledgement Number is 2000. The receiver also has enough buffer space for another 6400 bytes, so it has advertised a Window of 6000. The sender is currently sending a segment of 1000 bytes starting with Sequence Number 4001. The sender has received no acknowledgement for the bytes from 2001 on, but continues sending data as long as it is within the window. If the sender fills the window and receives no acknowledgement of the data previously sent, it will, after an appropriate time-out, resend the data starting from the first unacknowledged byte. Retransmission would start from byte 2001 if no further acknowledgements are received. This procedure ensures that data is reliably received at the far end of the network.

From the perspective of Applications, communication with the network involves sending and receiving continuous streams of data. It seems that the Application is not responsible for fragmenting the data to fit lower-layer protocols. The whole process can be illustrated in the following figure.

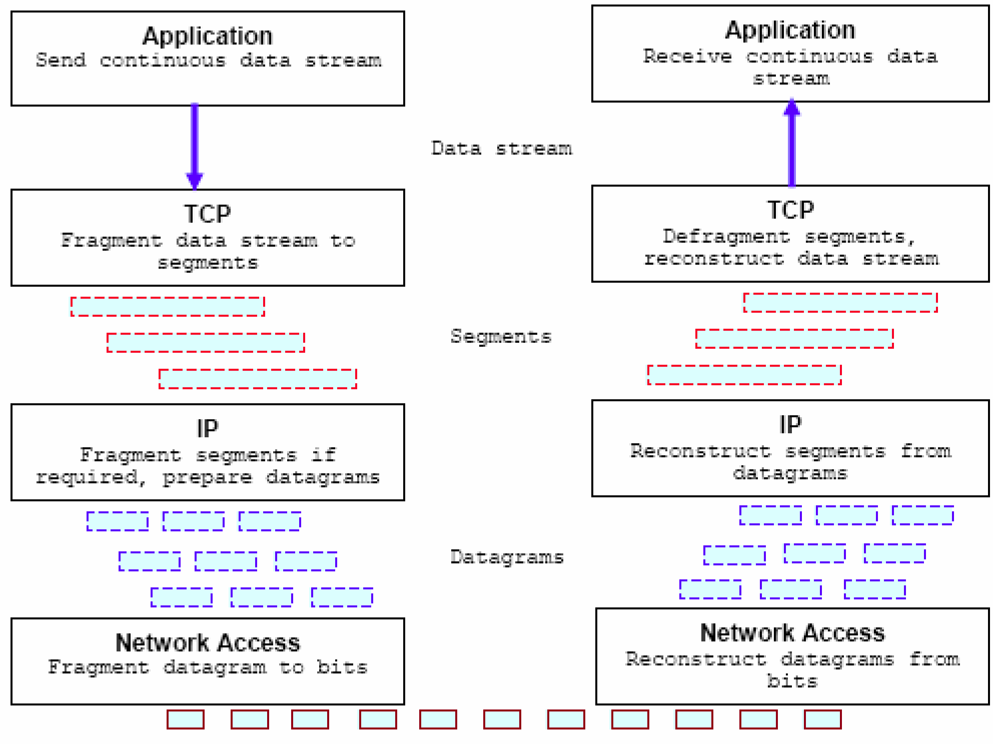

Application
Send continuous data stream
Application
Receive continuous data stream
Data stream
TCP
Fragment data stream to segments
TCP
Defragment segments, reconstruct data stream
Segments
IP
Fragment segments if required, prepare datagrams
IP
Reconstruct segments from datagrams
Datagrams
Network Access
Fragment datagram to bits
Network Access
Reconstruct datagrams from bits

Figure 22: How data is processed as they travel down the protocol stack, through the network, and up the protocol stack of the receiver.

Brief description:

- TCP receives a stream of data from the upper-layer process.
- TCP may fragment the data stream into segments that meet the maximum datagram size of IP.
- IP may fragment segments as it prepares datagrams that are sized to conform to restrictions of the network types: Ethernet, Token Ring etc.
- Network protocols transmit the datagram in the form of bits.
- Network protocols at the receiving host reconstruct datagrams from the bits they receive.
- IP receives datagrams from the network. Where necessary datagram fragments are reassembled to reconstruct the original segment.
- TCP presents data in segments to upper-layer protocols in the form of data streams.

Application Layer

The **Application** layer includes all processes that use the transport layer protocols to deliver data. There are many applications protocols. A good example of concerns handled by these processes is the reconciliation of differences in the data syntax between the platforms on which the applications are running. It should be clear that unless this difference in data representation is handled properly, any exchange of data involving these processes id likely to yield erroneous interpretations of numerical data. To resolve this issue, and other similar issues, TCP/IP defines the **eXternal Data Representation (XDR)** protocol. Reflecting on the nature of this problem, you can easily see that the problem has nothing to do with the underlying network topology, wiring, or electrical interference. Application examples that use TCP:

- TELNET: The Network Terminal Protocol provides remote login over the network.
- FTP: The File Transfer Protocol is used for interactive file transfer between hosts.
- SMTP: The Simple Mail Transfer Protocol acts as Mail Transfer Agent (MTA) that delivers electronic mail.

Application examples that use UDP:

- SNMP: The Simple Network Management Protocol is used to collect management information from network devices.
- DNS : Domain Name Service, maps IP addresses to the names assigned to network devices.
- RIP: Routing Information Protocol, routing is the central to the way TCP/IP networks. RIP is used by the network devices to exchange routing information.

- NFS : Network File System, this protocol allows files to be shared by various hosts on the network as if they were local drives.

RAW vs Cooked Socket

Intro

In this section and that follows, we will learn the basics of using raw sockets. Here, we will try to construct our own packet and insert any IP protocol based datagram into the network traffic. This is useful, for example, to build raw socket scanners like **nmap**, to **spoof** or to perform operations that need to send out raw sockets. Basically, you can send any packet at any time, whereas using the interface functions for your systems IP-stack (`connect()`, `write()`, `bind()`, etc.) as discussed in the previous Modules but you don't have direct control over the packets. This theoretically enables you to simulate the behavior of your OS's IP stack, and also to send stateless traffic (datagrams that don't belong to any valid connection).
The usage of the raw socket is to send a single packet at one time, with all the protocol headers filled in by the program (instead of the kernel). As discussed in the previous Modules, when you create a socket and bind it to a process/port, you don't care about IP or TCP header fields as long as you are able to communicate with the server. The kernel or the underlying operating system builds the packet including the checksum for your data. Thus, network programming was so easy with the traditional **cooked** sockets. Contrarily, raw sockets let you fabricate the header fields including information like source IP address etc. The following is a `socket()` prototype.

```
int socket(int domain, int type, int protocol);
```

If you check the man page for `socket()`, the socket types defined for the `type` parameter includes:

Type	Description
SOCK_STREAM	Provides sequenced, reliable, two-way, connection-based byte streams. An out-of-band data transmission mechanism may be supported.
SOCK_DGRAM	Supports datagrams (connectionless, unreliable messages of a fixed maximum length).
SOCK_SEQPACKET	Provides a sequenced, reliable, two-way connection-based data transmission path for datagrams of fixed maximum length; a consumer is required to read an entire packet with each read system call.
SOCK_RAW	Provides raw network protocol access.
SOCK_RDM	Provides a reliable datagram layer that does not guarantee ordering.
SOCK_PACKET	Obsolete and should not be used in new programs. Use packet (check man page for packet) instead.

Table 6: Socket types of `socket()`.

There are two methods of receiving packets from the datalink layer under Linux. The original method, which is more widely available but less flexible and obsolete, is to create a socket of type `SOCK_PACKET`. The newer method, which introduces more filtering and performance features, is to create a socket of family `PF_PACKET`. To do either, we must have sufficient privileges (similar to creating a raw socket), and the third argument to socket must be a nonzero value specifying the **Ethernet** (may use other frame type such as Token Ring etc.) frame type. When using `PF_PACKET` sockets, the second argument to socket can be `SOCK_DGRAM`, for "cooked" packets with the link-layer header removed, or `SOCK_RAW`, for the complete link-layer packet. `SOCK_PACKET` sockets only return the complete link layer packet. For example, to receive all frames from the datalink, we may write:

```
/* newer systems*/
fd = socket(PF_PACKET, SOCK_RAW, htons(ETH_P_ALL));
```

Or

```
/* older systems*/
fd = socket(AF_INET, SOCK_PACKET, htons(ETH_P_ALL));
```

This would return frames for all protocols that the datalink receives. If we want only IPv4 frames, the call would be:

```
/* newer systems */
fd = socket(PF_PACKET, SOCK_RAW, htons(ETH_P_IP));
```

Or

```
/* older systems */
fd = socket(AF_INET, SOCK_PACKET, htons(ETH_P_IP));
```

Other constants for the final argument are ETH_P_ARP and ETH_P_IPV6, for example.
And for the domain parameter constants are listed below. In the previous Modules we just use the AF_INET.

Name	Purpose
PF_UNIX, PF_LOCAL	Local communication.
PF_INET	IPv4 Internet protocols.
PF_INET6	IPv6 Internet protocols.
PF_IPX	IPX - Novell protocols.
PF_NETLINK	Kernel user interface device.
PF_X25	ITU-T X.25 / ISO-8208 protocol.
PF_AX25	Amateur radio AX.25 protocol.
PF_ATMPVC	Access to raw ATM PVCs.
PF_APPLETALK	Appletalk.
PF_PACKET	Low level packet interface.

Table 7: Domain parameters of socket().

The protocol parameter specifies a particular protocol number/name string to be used with the socket. Normally only a single protocol exists to support a particular socket type within a given protocol family, in which a case protocol can be specified as 0 (as used in the program examples in previous Modules). However, it is possible that many protocols may exist, in which case a particular protocol must be specified in this manner. The protocol number to use is specific to the communication domain. Partial list of the protocol number have been discussed in **IPv4 Datagram Format** section (or you can check man getprotobyname() page or RFC1700 for a complete list). To map protocol name strings to protocol numbers you may use getprotoent() function.
In the previous Modules we have already made familiar with SOCK_STREAM and SOCK_DGRAM. In this section, we'll be using SOCK_RAW, which includes the IP headers (and all subsequent protocol headers of the upper layer as needed) and data. In the previous program examples also we used SOCK_STREAM (TCP/connection oriented) and SOCK_DGRAM (UDP/connectionless) sockets as shown below:

```
socket(AF_INET, SOCK_STREAM, 0);
```

And

```
socket(AF_INET, SOCK_DGRAM, 0);
```

For raw socket we code as follows:

```
#include <sys/socket.h>
#include <netinet/in.h>

socket(PF_INET, SOCK_RAW, IPPROTO_UDP);
socket(PF_INET, SOCK_RAW, IPPROTO_TCP);
socket(PF_INET, SOCK_RAW, IPPROTO_ICMP);
```

Depending on what you want to send, you initially open a socket and give it its type. For example:

```
sockd = socket(AF_INET, SOCK_RAW, <protocol>);
```

For the <protocol>, you can choose from any protocol (number or string) including IPPROTO_RAW. The protocol number goes into the IP header verbatim. IPPROTO_RAW places 0 in the IP header. A socket option IP_HDRINCL allows you to include your own IP header along with the rest of the packet. Then, you might use it as:

```
char on = 1;
setsockopt(sockd, IPPROTO_IP, IP_HDRINCL, &on, sizeof(on));
```

You then build the packet and use a normal sendto(), recvfrom() etc.
The Internet IPv4 layer generates an IP header when sending a packet unless the IP_HDRINCL socket option is enabled on the socket. When it is enabled, the packet must contain an IP header that you should include in your

program. Only processes with an effective user id of 0 (root) or the CAP_NET_RAW capability are allowed to open raw sockets. A protocol of IPPROTO_RAW implies the IP_HDRINCL is enabled. For this case, the following is a summary for the IP header.

IP Header fields modified on sending by IP_HDRINCL

IP Checksum	Always filled in.
Source Address	Filled in when zero.
Packet Id	Filled in when zero.
Total Length	Always filled in.

Table 8: Some IP header default values.

If IP_HDRINCL is specified and the IP header has a non-zero destination address then the destination address of the socket is used to route the packet. When MSG_DONTROUTE is specified the destination address should refer to a local interface, otherwise a routing table lookup is done anyways but gatewayed routes are ignored. If IP_HDRINCL isn't set then IP header options can be set on raw sockets with setsockopt() as shown before. Raw sockets are usually only needed for new protocols or protocols with no user interface (like ICMP). When a packet is received, it is passed to any raw sockets which have been bound to its protocol before it is passed to other protocol handlers (e.g. kernel protocol modules).

Raw sockets use the standard sockaddr_in address structure defined in ip.h. For example:

```
...
struct sockaddr_in sin;
...
// Address family
sin.sin_family = AF_INET;
// Port numbers
sin.sin_port = srcportnum;
// IP addresses
sin.sin_addr.s_addr = inet_addr(argv[1]);
```

Raw socket options can be set with setsockopt() and read with getsockopt() by passing the SOL_RAW family flag. Raw sockets fragment a packet when its total length exceeds the interface **Maximum Transfer Unit (MTU)**. A raw socket can be bound to a specific local address using the bind() call. If it isn't bound, all packets with the specified IP protocol are received. In addition a RAW socket can be bound to a specific network device using SO_BINDTODEVICE (check the socket() man page).

An IPPROTO_RAW socket is send only. If you really want to receive all IP packets use a packet() socket with the ETH_P_IP protocol. Note that packet sockets don't reassemble IP fragments, unlike raw sockets.

If you want to receive all ICMP packets for a datagram socket it is often better to use IP_RECVERR on that particular socket.

Raw sockets may tap all IP protocols in Linux for example, even protocols like ICMP or TCP which have a protocol module in the kernel. In this case the packets are passed to both the kernel module and the raw socket(s).

--Break--

Note:

The Maximum Transfer Unit (MTU) specifies the maximum transmission unit size of an interface. Each interface used by TCP/IP may have a different MTU value specified. The MTU is usually determined through negotiation with the lower-level driver and by using that lower-level driver value. However, that value may be overridden. Each media type (used in Ethernet, FDDI, Token Ring etc) has a maximum frame size that cannot be exceeded. The link layer is responsible for discovering this MTU and reporting it to the protocols above the link layer. Network Driver Interface Specification (NDIS) drivers may be queried for the local MTU by the protocol stack. Knowledge of the MTU for an interface is used by upper-layer protocols, such as TCP, which automatically optimizes packet sizes for each medium.

--Break--

From the moment the raw socket is created, you can send any IP packets over it, and receive any IP packets that the host received after that socket was created if you read() from it. Note that even though the socket is an interface to the IP header, it is transport layer specific. That means, for listening to TCP, UDP and ICMP traffic, you have to create 3 separate raw sockets, using IPPROTO_TCP, IPPROTO_UDP and IPPROTO_ICMP (the protocol numbers are 6 for tcp, 17 for udp and 1 for icmp).

With this knowledge, we can, for example, create a small **sniffer program** as shown in the following code portion that dumps out the contents of all tcp packets we receive and print out the payload, the data of the session/application layer etc.

```
int fd = socket(PF_INET, SOCK_RAW, IPPROTO_TCP);
/* single packets are usually not bigger than 8192 bytes but
   depend on the media standard of the Network Access layer such as
   Ethernet, Token Ring etc
*/

...
char buffer[8192];
struct ipheader *ip = (struct ipheader *) buffer;
struct udpheader *udp = (struct udpheader *) (buffer + sizeof(struct ipheader));

...
while (read(fd, buffer, 8192) > 0)
/* packet =  data + ip header + tcp header */
/* Little Endian/Big Endian must be considered here */
 printf("Dump the packet: %s\n", buffer + sizeof(struct ipheader) + sizeof(struct tcpheader));
```

Continue on next Module…TCP/IP and RAW socket, more program examples.

-------------------------------------End Part IV-------------------------------------

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Further reading and digging:

1. Check the best selling C/C++, Networking, Linux and Open Source books at Amazon.com.