Computer Networks 计算机网络

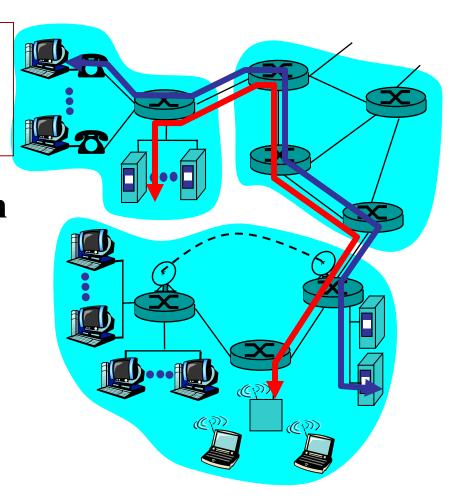
Classification of Networks

- by scale
 LAN, WAN and MAN
- by topology
 Bus, Star, Ring and Tree
- by switching approach
 Circuit switching and Packet switching
- **by transmission media**Wireless network and Wired network

Circuit Switching电路交换

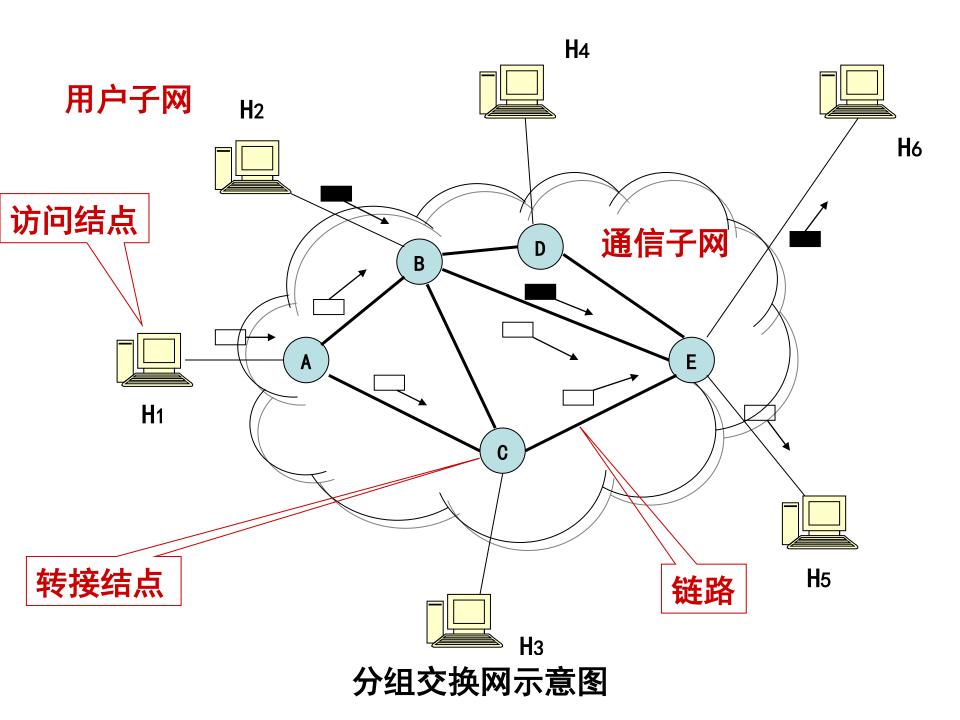
End-to-end resources reserved for "call"

- ≥ Link bandwidth, switch capacity
- **➣** Dedicated resources with no sharing
- **™** Guaranteed transmission capacity
- **™** Call setup required

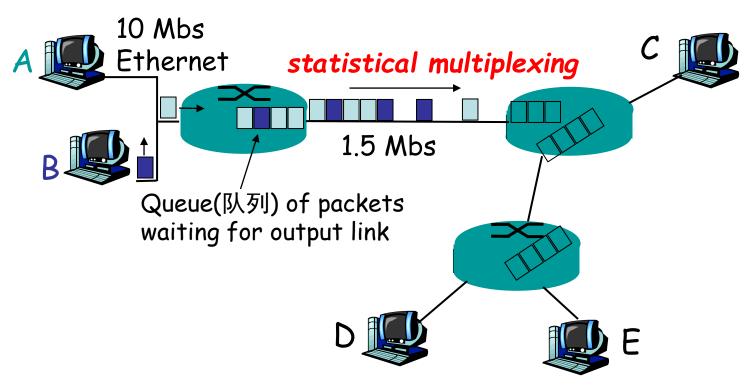


Packet Switching分组交换

- osource breaks long messages into smaller "packets(数据包,数据分组)"
- packets share network resources
- each packet uses full link bandwidth
- "store-and-forward(存储转发)" transmission



Packet Switching: Statistical Multiplexing



Sequence of A & B packets does not have fixed pattern □ **statistical multiplexing统计多路复**用.

Four sources of packet delay

- 1. d_{proc} = processing delay(处理时延) typically a few microsecs or less
- 2. d_{queue} = queuing delay(排队时延) depends on congestion
- 3. d_{trans} = transmission delay(发送时延) = L/R, significant for low-speed links
- 4. d_{prop} = propagation delay(传播时延) a few microsecs to hundreds of msecs

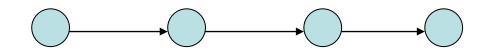
Nodal delay

$$d_{\text{nodal}} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{trans}} + d_{\text{prop}}$$

总时延=处理时延+排队时延+发送时延+传播时延

• 补充题: 试在下列条件下比较电路交换和分组 交换。要传送的报文共x (bit),从源站到目的站 共经过 k 段链路,每段链路的传播时延为d 秒, 数据率为 b (bit/s)。在电路交换时电路的建立时 间为 s 秒。在分组交换时分组长度为 p (bit),且 各结点的排队等待时间可忽略不计。问在怎样 的条件下,分组交换的时延比电路交换要小?

补充题:



- 电路交换时延: s+x/b+kd
- 分组交换时延: x/b+kd+(k-1)p/b

$$x/b+kd+(k-1)p/b < s+x/b+kd$$

$$\rightarrow$$
 s > (k-1)p/b

*但前提是: x>>p,或分组数大于链路数.

• 补充题2: 在上题的分组交换网中,设报文和 分组长度分别为x和(p+h)(bit),其中p为分 组的数据部分的长度,而 h 为每个分组所带的 控制信息固定长度,与p的大小无关。通信的 两端共经过 k 段链路。链路的数据率为b(bit/s), 但传播时延和结点的排队时延均可忽略不计。 若打算使总的时延为最小。问分组的数据部分 长度 p 应该取多大?

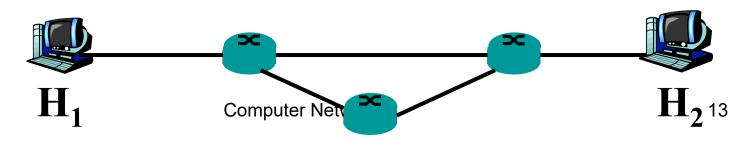
补充题 (续)

- 分组数: $\frac{x}{p}$, 发送的数据量: $x + \frac{x}{p} \cdot h$
- 总时延 $D = \left(x + \frac{x}{p} \cdot h\right) \cdot \frac{1}{b} + (k-1) \cdot \frac{(p+h)}{b}$
- 求D对p的导数,令其为0: $D'_p = 0$

$$\implies p = \sqrt{xh/(k-1)}$$

2010年全国硕士研究生入学统一考试 计算机学科专业基础综合

在下图所示的采用"存储-转发"方式的分 组交换网络中,所有链路的数据传输速率为 100Mbps, 分组大小为1000 B, 其中分组头 大小为20 B。若主机H1向主机H2发送一个大 小为980 000 B的文件,则在不考虑分组拆装 时间和传播延迟的情况下,从H1发送开始到 H2接收完为止,至少需要多少时间?



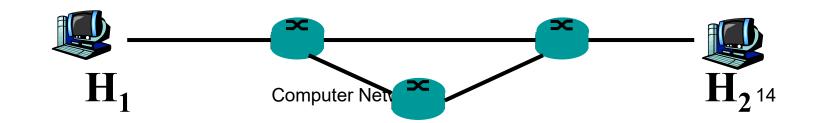
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$$B=100Mbps; x=980\ 000\ B$$

$$p=(1000-20) B; h=20B; k=2$$

$$D = \left(x + \frac{x}{p} \cdot h\right) \cdot \frac{1}{b} + (k-1) \cdot \frac{(p+h)}{b}$$

$$\mathbf{D=80.16 \ msec}$$



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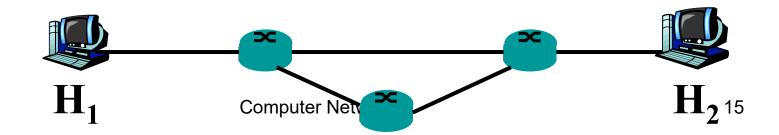
分组长度L=1000B=980B+20B

分组数N=980000/980=1000

发送1个分组的时间T_{tran}=(1000x8)/(100x10⁶)

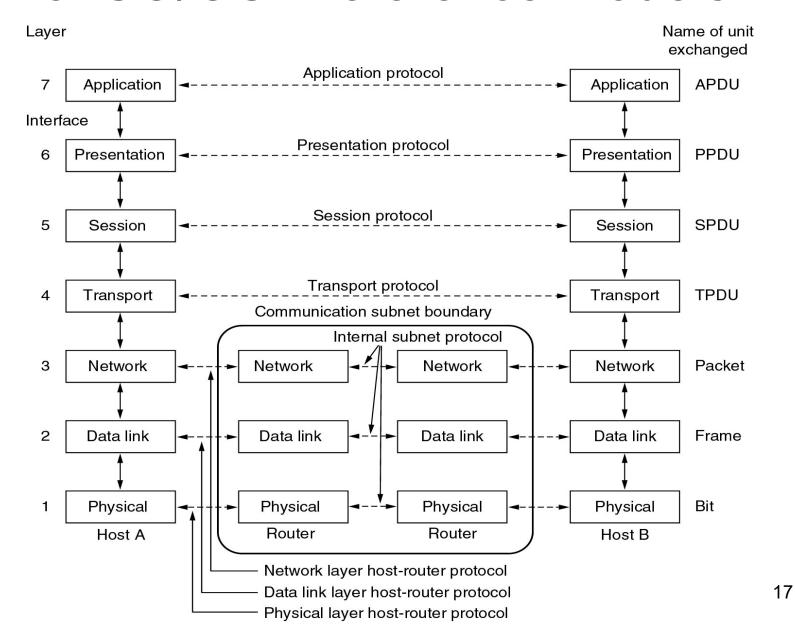
 $=8x10^{-5}$ sec

 $T_{total} = N \times T_{tran} + 2 \times T_{tran} = 80.16 \text{ msec}$



- OSI (Open System Interconnection) is an ISO standard for worldwide communications
- The OSI Reference Model defines a networking framework for implementing protocols in seven layers.

The ISO/OSI Reference Models

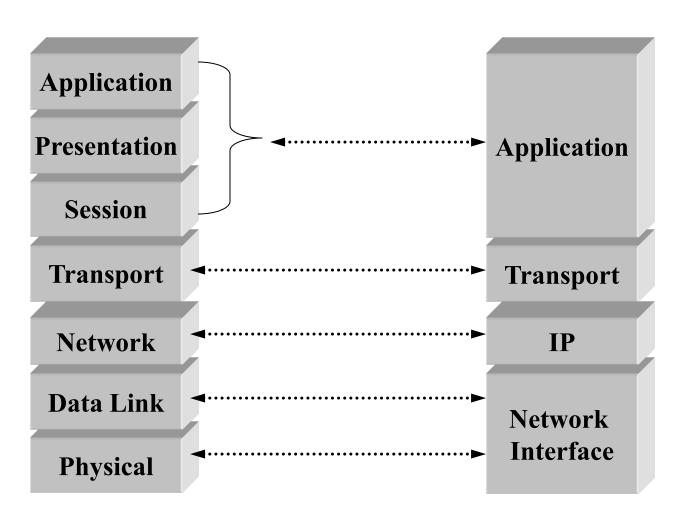


- ① Physical Layer → how to transmit bits to the channel;
- ② Data Link Layer → how to transmit frames to adjacent node (neighbour), over a single link;
- Network Layer → how to route packets to a host on the other side, across network(s)

- Transport Layer → how to send data segments to a process running on another host, across network(s)
- Session Layer → manage connections
- 6 Presentation Layer → encode/decode messages, security, encryption

- Transport Layer → how to send data segments to a process running on another host, across network(s)
- ⑤ Session Layer → manage connections
- 6 Presentation Layer → encode/decode messages, security, encryption

Comparison: OSI and TCP/IP



The Physical Layer

- Mechanical and electrical specifications
- Encoding/Dencoding techniques
- **Propagation Effects:**
 - Attenuation衰减,
 - Distortion失真,
 - Noise噪音,
 - Interference冲突, ...

The Physical Layer

- Bandwidth带宽: Capacity of a media to carry information
- A channel is a portion of the total bandwidth used for a specific purpose.
 - Simplex channel 单工信道,
 - Half-duplex channel 半双工信道
 - Full-duplex channel 全双工信道.



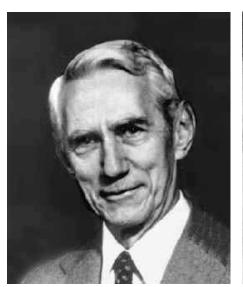
Harry Nyquist (1889--1976), an important contributor to information theory.

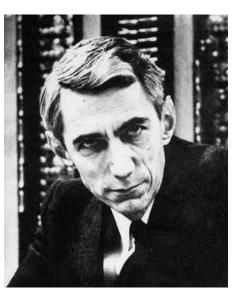
⇒如果一个任意信号通过带宽为H的低通滤波器,那么只需要每秒采样2H次就能完全地重现被滤波的信号。

Suppose we know the bandwidth (H) of a channel and the number of signal levels (V) being used. What is the maximum number of bit we could transmit?

Nyquist's theorem says:

Max data rate
$$R_{\text{max}} = 2H \log_2 V$$
 bits/sec





Claude Elwood Shannon 1916 - 2001

In 1948, he published a research paper at Bell System Technical Journal: "通信中的数学原理".

⇒ Shannon's major result is that the maximum data rate of a noisy channel whose bandwidth is *H* Hz, and whose signal-to-noise ratio is *S/N*, is given by

$$R_{\text{max}} = H \log_2(1 + S/N)$$
 bits/sec

⇒Shannon's result applies to any channel subject to thermal noise香农的结论适用于任何带有热噪声信道.

The Data Link Layer

- The data link layer is responsible for efficient reliable communication across a physical link.
 - **⇔** LOGICAL LINK sublayer (LLC)
 - MEDIA ACCESS sublayer (MAC)
 - Ethernet (IEEE802.3)
 - Wireless LAN (IEEE802.11)

The Data Link Layer

OSI

Application

Presentation

Session

Transport

Network

Data Link

Physical

LOGICAL LINK sublayer

Framing

Error control

Flow control

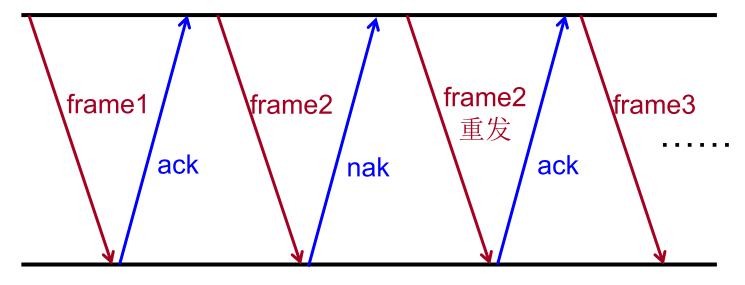
MEDIA ACCESS sublayer

Transmission/reception of frames

Data Link Protocols

Protocols in which the sender sends one frame and then waits for an acknowledgement before proceeding are called stop-and-wait.

Sender A



Receiver B

Problem: B sends a NAK frame back to A, after having received a data frame with errors. What happens if A always gets NAK frames.

Solution: set a max. number for retransmission times, *e.g.* 8. If not successful, gives an error report to the above layer.

Problem: Due to poor link conditions, the frame sent by A doesn't get B at all. It gets lost! In this case, A will never get any response from the peer.

Solution: schedule a timeout timer to expire at some time after the ACK should have been returned. If the timer goes off, retransmit the frame.

Problem: Retransmissions may introduce duplicate frames received by B

Solution: assign sequence numbers序号 for every frame, so that B can distinguish between new frames and old copies. However, an ACK for the duplicated frame is still necessary!

- A protocol, in which the sender waits for a positive acknowledgement before advancing to the next data item, are often called ARQ (Automatic Repeat reQuest).
- The ARQ protocol is very simple;
- **⊃**Unfortunately, it gives poor link utilization.



t prop → Propagation delay:

- defined as the delay between transmission and receipt of frames between hosts
- can be used to estimate timeout period

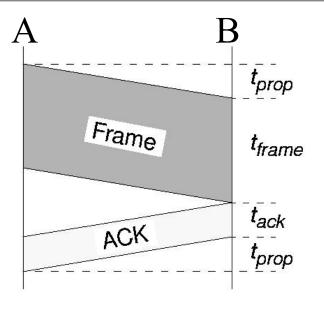
$$t_{frame}$$

→ Frame transmission time

 $t_{ack} \rightarrow$ Acknowledgment transmission time

 $T_D \rightarrow \text{Total delay (ignoring ACK transmission time):}$

$$T_D = 2t_{prop} + t_{frame}$$





Stop-and-Wait Protocol Performance

Of this time, only t_{frame} is actually spent transmitting data. Therefore, the efficiency or utilization is:

$$U = rac{t_{\mathit{frame}}}{T_{\!D}} = rac{t_{\mathit{frame}}}{2t_{\mathit{prop}} + t_{\mathit{frame}}}$$

If we define a:

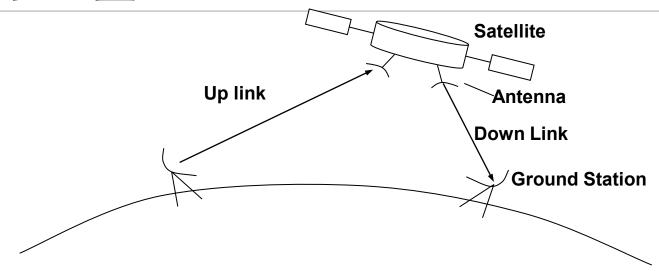
$$a = \frac{t_{prop}}{t_{frame}}$$

then

$$U = \frac{1}{1 + 2a}$$

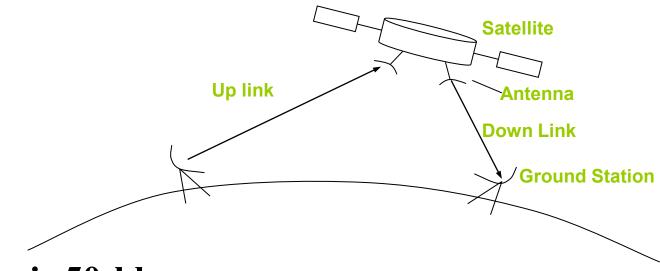


Quiz





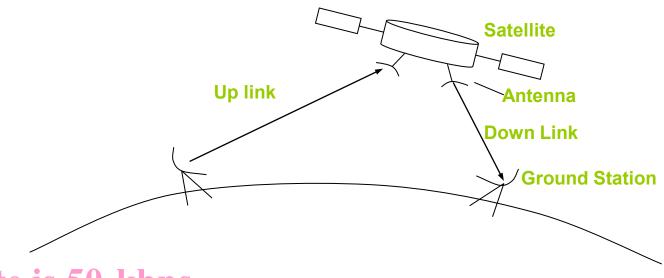
Quiz



- □ The data rate is 50-kbps.
- \square The round-trip delay is 500ms.
- □ What is the link utilization, if you use stop-and-wait protocol to send 1000-bit frames?



Long Transit Delays

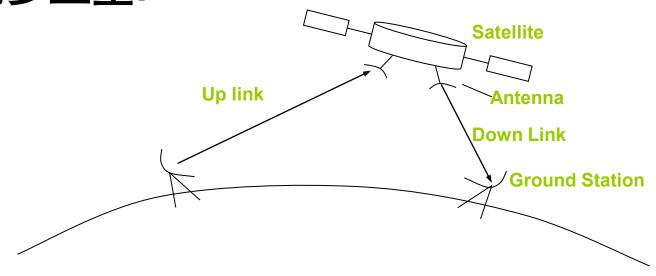


- □ The data rate is 50-kbps.
- □ The round-trip delay is 500 msec.
- ☐ If we use stop-and-wait protocol to send a 1000-bit frame, the receiver will get the whole frame 270msec



Long Transit Delays

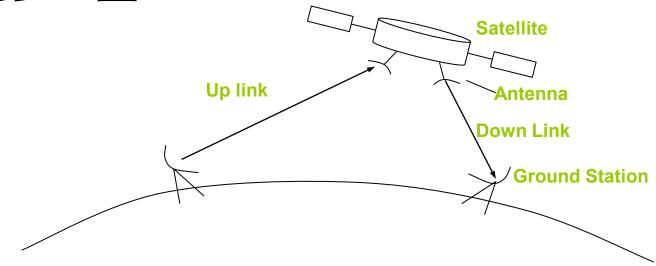
Imagine a link that uses a geo-stationary satellite地球同步卫星:



□ The acknowledgement will take a further 250*msec* to get back.



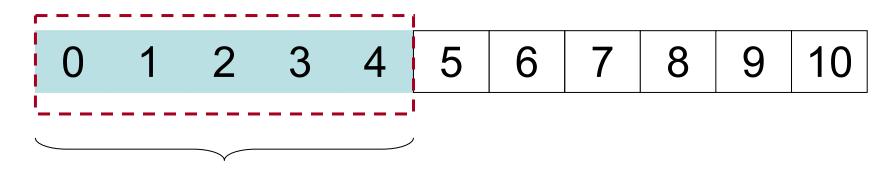
Long Transit Delays



- □ Out of 520*msec*, data is only being sent for 20*msec*.
- □ Only $20/520 \approx 4\%$ of the link's capacity is being utilised.

滑动窗口协议 (Sliding Window Protocols)

- · 目的: 限制发送方已经发出,但未被确认的数据帧的数目。
- 发送窗口用来控制发送方的流量。发送窗口内的帧是允许发送的帧,而不考虑有没有收到接收方的确认。



数据链路层协议

- Go Back n后退n帧协议: to discard the damage frame and all the frames that follow it, then retransmit all of them.
- Selective Repeat选择性重传: With selective repeat, only those frames that are damaged are re-sent.

- The 802.3 standard describes the operation of the MAC sub-layer in a bus LAN that uses CSMA/CD
- Mechanism for Channel Access信道访问机制 CSMA/CD: Carrier Sense Multiple Access with Collision Detection 载波侦听多路存取/碰撞检测

CSMA载波侦听、多路存取

- Carrier Sense: With carrier sensing, A host will only transmit its own frames when it cannot hear any data being transmitted by other hosts.
- Multiple Access: Multiple hosts share a single channel

载波侦听:每个主机在发送数据之前首先监听信道,只有当信道空闲时才发送数据帧;如果信道忙,暂不发送(退避一个随机的时间),以免发生碰撞;

多路存取: 多个主机共享信道。

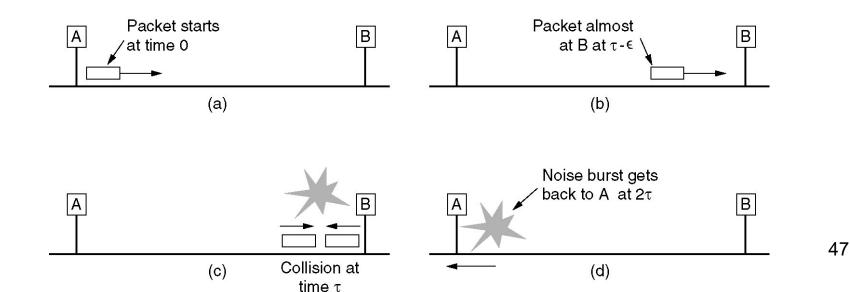
CSMA/CD 载波侦听多路存取/冲突检测

CSMA with Collision Detection

- ➣ The host always listens to the cable while it is transmitting data.
- ≥ It aborts transmission as soon as it detects a collision

冲突检测: 在发送数据的过程中始终监听信道, 一旦冲突立即中止发送, 退避一个随机时间(二进制指数退避算法), 再重新发送。

- When a host transmits a frame, there is a small chance that a collision will occur, i.e. non-deterministic不确定性
- The frame should be longer enough for sender to detect the collision



- This means that the frame must be of a minimum length.
- The minimum frame size is related to
 - the distance which the network spans;
 - the type of media being used;
 - ≥ the number of repeaters中继器 which the signal may have to pass through to reach the furthest part of the LAN.

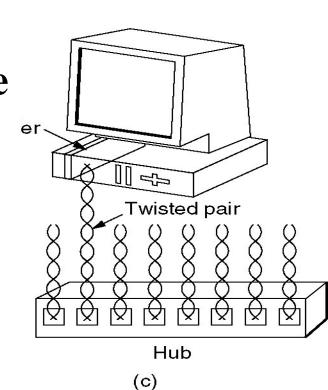
• Ethernet defines a minimum frame size, i.e. no frame may have less than 46 bytes of payload.

Minimum Frame Length

- Two nodes are communicating using CSMA/CD protocol. Transmission rate is 100 Mbits/sec and frame size is 1500 bytes. The propagation speed is 3*10⁸ m/sec.
- Calculate the distance between the nodes such that the time to transmit the frame = time to recognize that the collision have occurred.

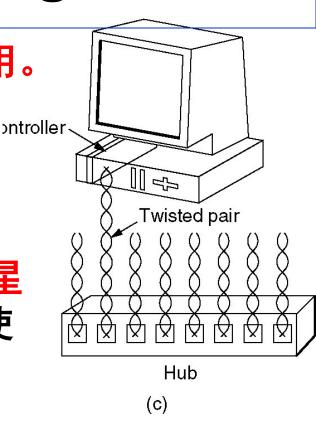
Ethernet Cabling

- In option of 10Base-T Ethernet Cabling, all stations have a cable running to a central hub 集线器 in which they are all connected electrically.
- In this case, the physical topology of the LAN is star, however, the logical topology is still bus.



Ethernet Cabling

- ·双绞线Ethernet总是和集线器配合使用。
- •术语: 10Base-T Ethernet
 - ▶"10"代表10Mbit/s的数据率。
 - > "T"代表双绞线星型网。
- ·用集线器Hub来连接站点。物理上是星型网,但逻辑上仍是总线网。各站仍使用CSMA/CD协议,并共享逻辑上的总线。



·各站必须竞争公共信道,并在任何时刻只有一个站可以 发送数据。否则发生碰撞!!

Ethernet Extension

Application layer	Application gateway
Transport layer	Transport gateway
Network layer	Router
Data link layer	Bridge, switch
Physical layer	Repeater, hub
,	(a)

- At the Physical Layer
 - Hub集线器
 - Repeaters中继器
- At the Data Link Layer
 - Bridge(网桥)
 - Switch(交换机)
- At the Network Layer
 - Router(路由器)



Wireless LAN无线局域网

■ IEEE 802.11 defines CSMA/CA protocol.

CSMA part is the same as in 802.3 Ethernet, CA stands for Collision Avoidance 冲突避免

How CSMA/CA works:

- Device wanting to transmit senses the medium (Air)
- If medium is busy defers
- If medium is free for certain period, transmits frame

CSMA/CA 载波监听多路存取/冲突避免

- avoid collisions: two or more nodes transmitting at same time
- ■802.11: CSMA sense(监听) before transmitting
 - don't collide with ongoing transmission by other node



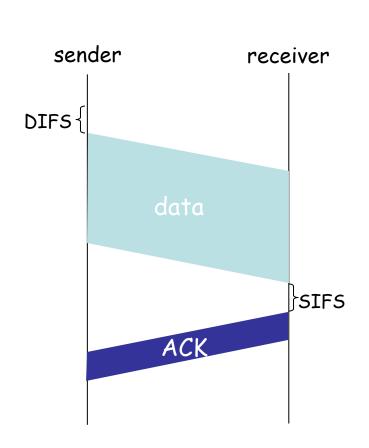
CSMA/CA 载波监听多路存取/冲突避免

802.11 sender

if sense channel idle for DIFS then transmit entire frame (no CD)

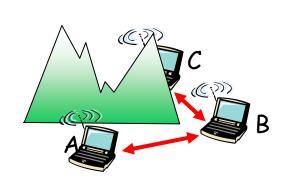
802.11 receiver

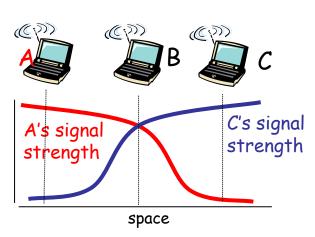
if frame received OK
 return ACK after SIFS (ACK
 needed due to hidden
 terminal problem)



CSMA/CA 载波监听多路存取/冲突避免

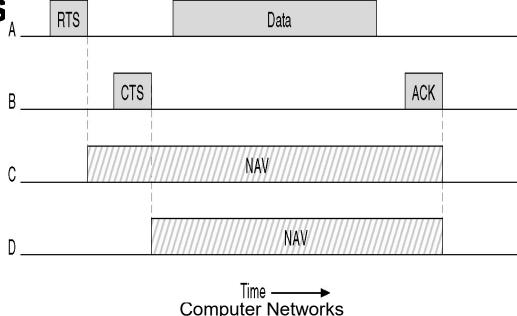
- ■802.11: no collision detection(没有冲突检测)!
 - can't sense all collisions in any case: hidden terminal隐蔽终端, fading信号衰减
 - goal: avoid collisions:CSMA/C(ollision)A(voidance)



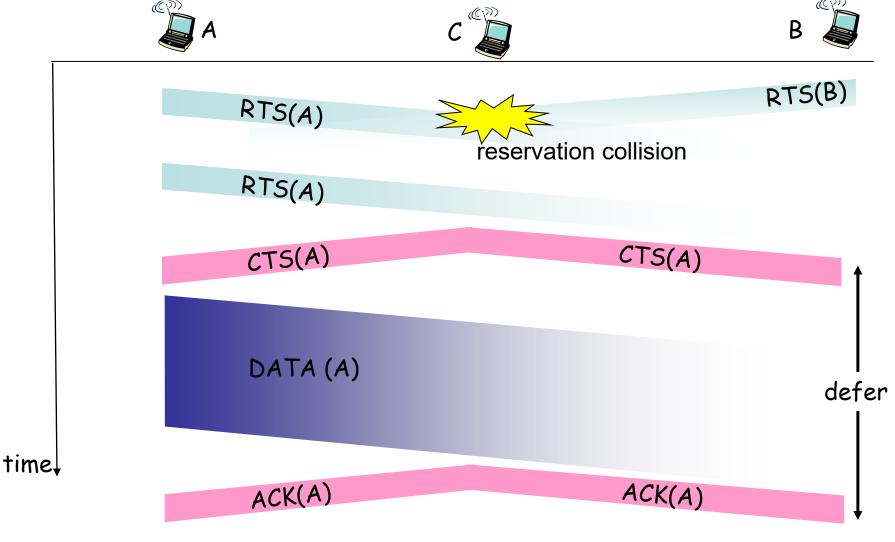


Wireless LAN无线局域网

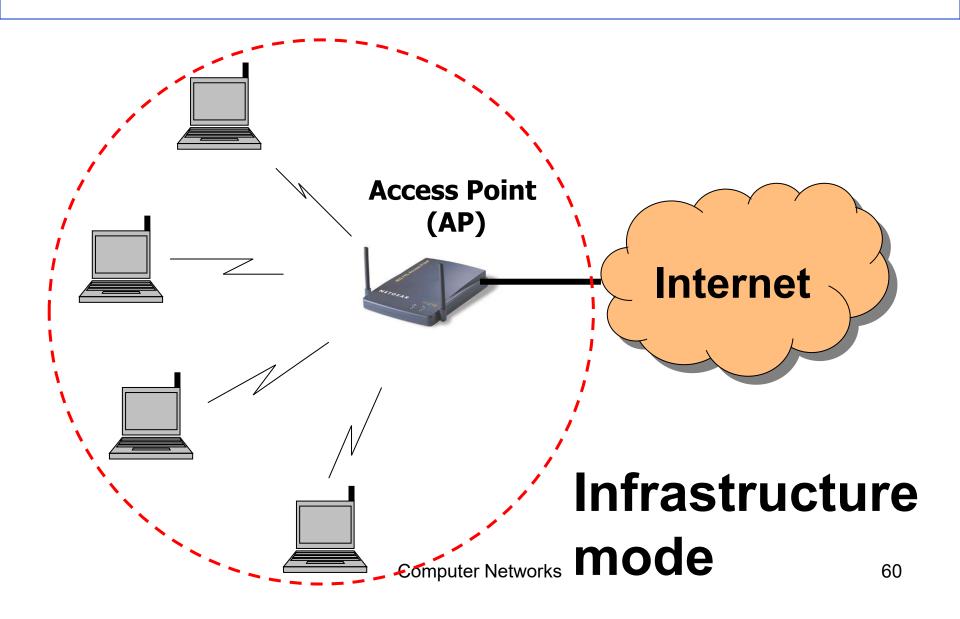
• By sending Request to Send (RTS), sender is allowed to "reserve" channel rather than random access of data frames: avoid collisions of long data frames.



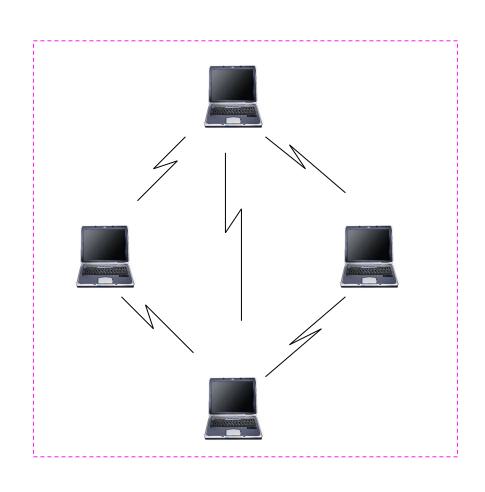
Collision Avoidance(冲突避免): RTS/CTS exchange



IEEE802.11 WLAN



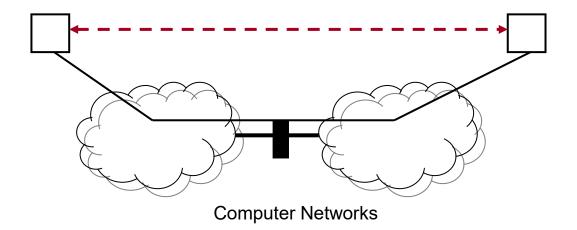
IEEE802.11 WLAN



Ad Hoc mode

The Network Layer

The network layer, based on services provided by the data link layer, provides a end-to-end transparent path for end-toend transparent data transmission across networks.



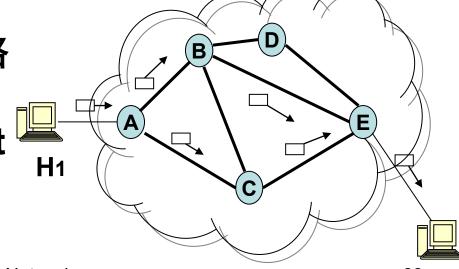
The functions of network layer:

Routing路由选择

packets are injected into the subnet individually and routed independently of each other.

Routing involves the selection of the best paths for packets from source to destination

The routing algorithm路 由选择算法 is responsible for deciding which output line an incoming packet should be transmitted on



IP Addresses

- Every node on the Internet has IP address(es)
- IP address is used to identify the network and the host on a given network
- Each IP address is 32 bits long, e.g. 10000000 00001011 00000011 00011111
- The IP address is divided into two parts:

```
IPaddr ::= { <net-id>, <host-id>}
```

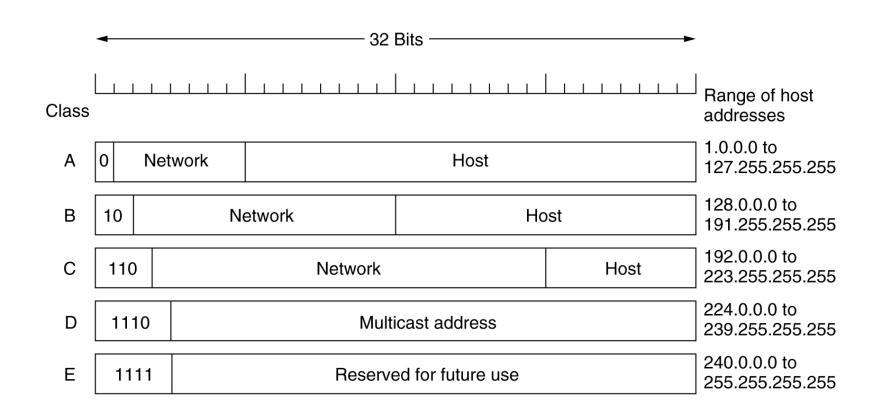
IP Addresses

● IP addresses are usually written in dotted decimal notation(点分十进制记法), e.g.

10000000 00001011 00000011 00011111

 $\Box \rightarrow 128.11.3.31$

Classes of IP Addresses IP地址分类



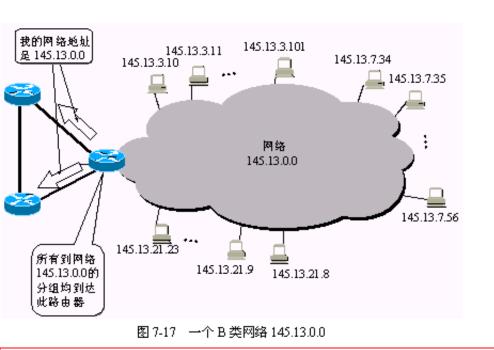
Subnetting子网划分

Basic idea:

- A organization with a large network can be divided to many smaller networks, i.e. subnets. Subnetting is an intramural matter.
- take some bits from the host number part to create a "subnet" number.

IPaddr ::=
$$\{ < net-id >, < host-id > \} \rightarrow$$
IPaddr ::= $\{ < net-id >, < subnet-id >, < host-id > \}$
Computer Networks

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A organization with a large network can be divided to many smaller networks, i.e. subnets. Subnetting is an intramural matter.

Subnetting: 145.13.3.11 145.13.3.101 所有目的蜥址为 145.13.7.34 [145.13.3.10] take some bits from the host 145.13.7.35 子阿 145.13.3.0 number part to create a 子网 145.13.7.0 "subnet" number. 子阿 145.13.21.0 145.13.7.56 IPaddr ::= {<net-id>, <subnet-id>, <host-id>} 145.13.21.23 网络 145.13.21.9 _{145.13.21.8} 145,13,0,0 Compu

Subnetting子网划分

- Packet routing from the source to the destination across the network:
 - > Destination Network
 - → Destination Subnet
 - → Destination Host
- Subnet masks子网掩码 indicates which part of a 32-bit IP address represents net-id and subnet-id

Subnetting子网划分

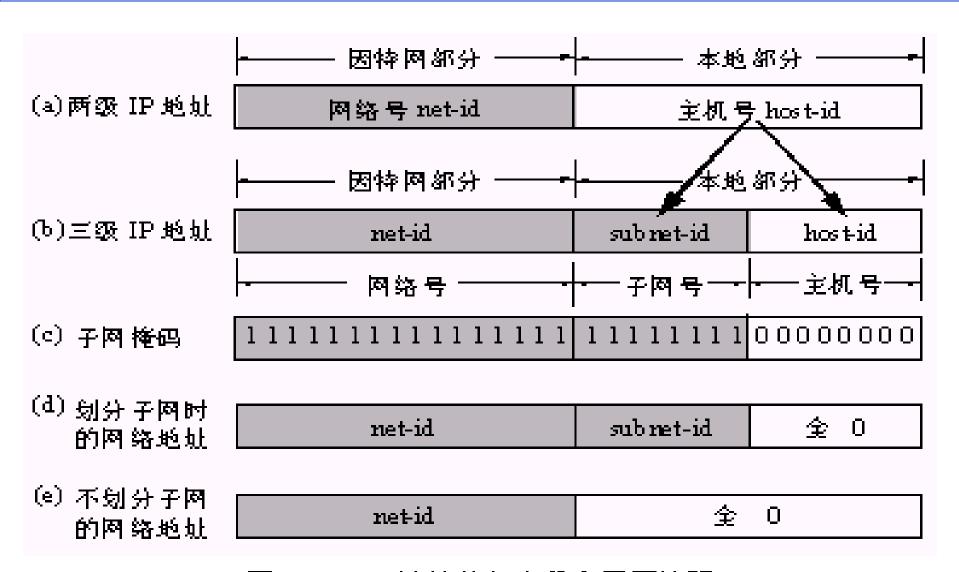


图 7-19 IP 地址的各字段和子网掩码

Subnet Examples

■ IP Address: 130.97.16.132

Subnet Mask: 255.255.255.192

- Net-id=?
- Host-id=?
- IP Address: 130.97.17.132

Subnet Mask: 255.255.254.0

- Net-id=?
- Host-id=?

CIDR – Classless InterDomain Routing 无类域间路由选择

net-id and host-id are replaced by network-prefix网络前缀

```
i.e. IPaddr ::= {<net-prefix>, <host-id>}. e.g. 128.14.46.34/20
```

→ 10000000 00001110 00101110 00100010

net-prefix host-id

常用的 CIDR 地址块

	• • • • • • • • • • • • • • • • • • • •	_ , ,	
CIDR 前缀长度	点分十进制	包含的地址数	包含的分类的网络数
/13	255.248.0.0	512 K	8 个 B 类或 2048 个 C 类
/14	255.252.0.0	256 K	4 个 B 类或 1024 个 C 类
/15	255.254.0.0	128 K	2 个 B 类或 512 个 C 类
/16	255.255.0.0	64 K	1 个 B 类或 256 个 C 类
/17	255.255.128.0	32 K	128个 C 类
/18	255.255.192.0	16 K	64 个 C 类
/19	255.255.224.0	8 K	32 个 C 类
/20	255.255.240.0	4 K	16 个 C 类
/21	255.255.248.0	2 K	8个C类
/22	255.255.252.0	1 K	4个C类
/23	255.255.254.0	512	2个C类
/24	255.255.255.0	256	1个C类
/25	255.255.255.128	128	1/2 个 C 类
/26	255.255.255.192	64	1/4 个 C 类
/27	255.255.255.224	32	1/8个 C 类

Chapter 5, Problem 27

A large number of consecutive IP address are available starting at 198.16.0.0. Suppose that four organizations, A, B, C, and D, request 4000, 2000, 4000, and 8000 addresses, respectively, and in that order. For each of these, give the first IP address assigned, the last IP address assigned, and the mask in the w.x.y.z/s notation.

Chapter 5, Solution 27

To start with, all the requests are rounded up to a power of two.

Organizations A, B, C and D want to have 4000, 2000, 4000, and 8000 addresses, respectively,

so the address for them must have a host-id of 12, 11, 12 and 13 bits long, the net-prefix is 20, 21, 20, 19 bits long, respectively.

- 单位A需要4000个地址,因此主机地址应为 12位(2¹²=4096);由于4000/256=15.625,因 此,末尾地址为198.16.15.255,即地址范围 为198.16.0.0—198.16.15.255
- · 单位B需要2000个地址, 主机地址11位, 2000/256=7.8125, 则地址范围为 186.16.16.0—198.16.23.255
- ·以此类推,可以得到单位C和D的地址范围
- · 如果单位B需要8000个地址,则主机地址需要13位,8000/256=31.25,则地址范围变成198.16.32.0—198.16.63.255

Chapter 5, Solution 27

Therefore, The starting address, ending address, and mask are as follows:

- A: 198.16.0.0 –198.16.15.255 written as 198.16.0.0/20
- B: 198.16.16. 0--198.16.23.255 written as 198.16.16. 0/21
- C: 198.16.32. 0--198.16.47.255 written as 198.16.32. 0/20;
- D: 198.16.64. 0-198.16.95.255 written as 198.16.64. 0/19

2010年全国硕士研究生入学统一考试 计算机学科专业基础综合



37. 某网络的IP地址空间为192.168.5.0/24,采用定长子网划分,子网掩码为255.255.255.248,则该网络中的最大子网个数、每个子网内的最大可分配地址个数分别是多少?



2010年全国硕士研究生入学统一考试 计算机学科专业基础综合



IP地址空间为192.168.5.0/24

XXXXXXX XXXXXXX XXXXXXX XXXXXXX

子网掩码为255.255.255.248

11111111 11111111 11111000 11111111



2010年全国硕士研究生入学统一考试 计算机学科专业基础综合

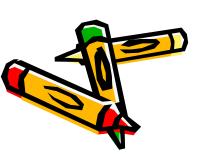
37. 单位网络的IP地址空间为192.168.5.0/24,

单位内部采用定长子网划分,

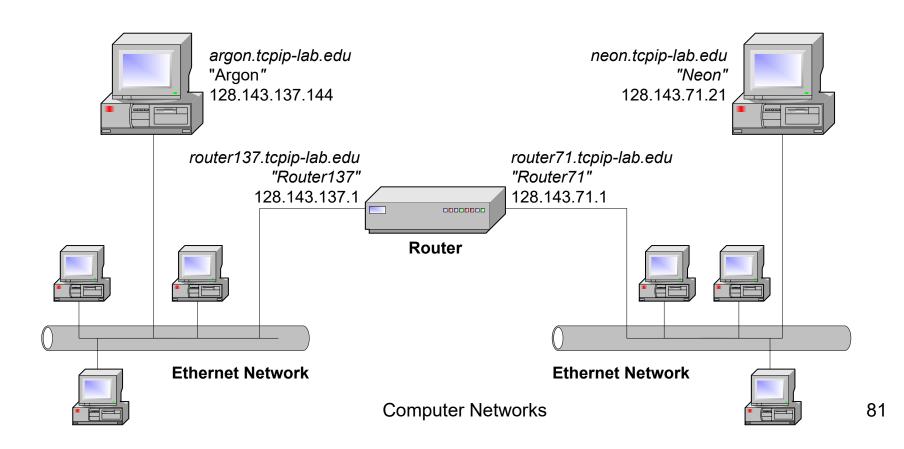
子网掩码为255.255.255.248,则

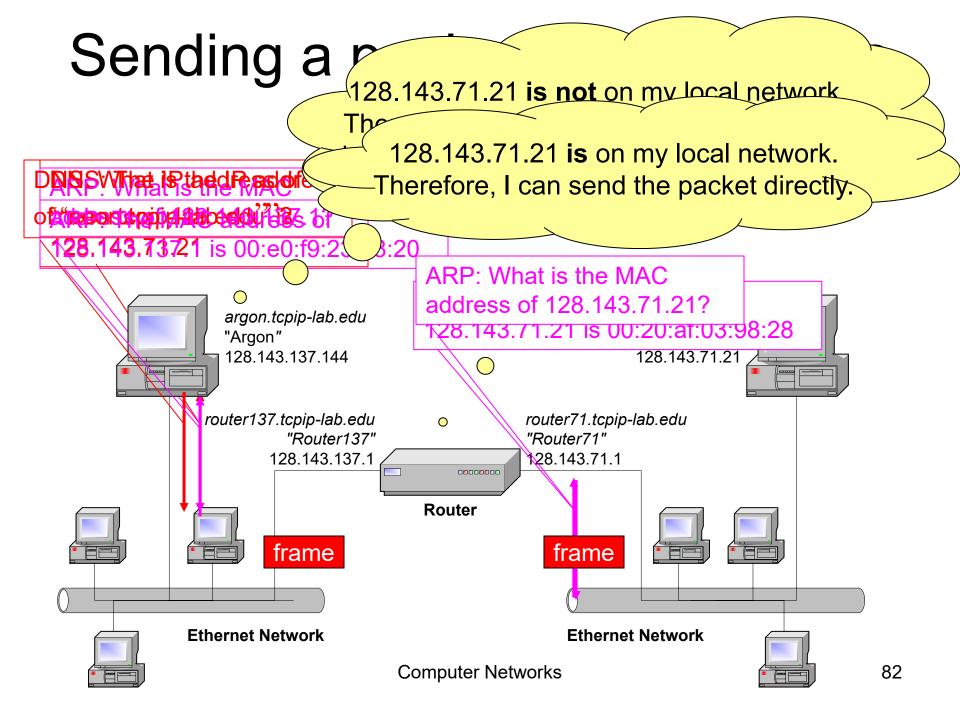
该网络中的最大子网个数: 32

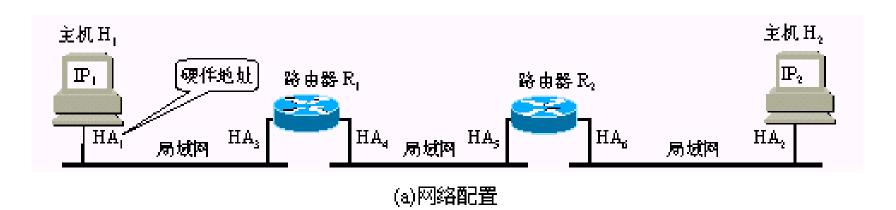
每个子网内的最大可分配地址个数是: 6

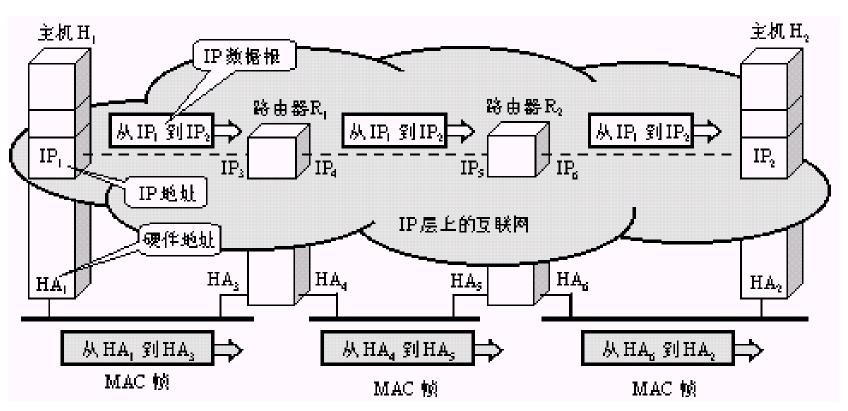


Sending a packet from Argon to Neon



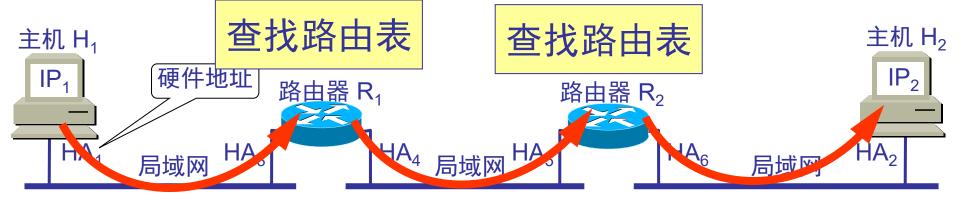






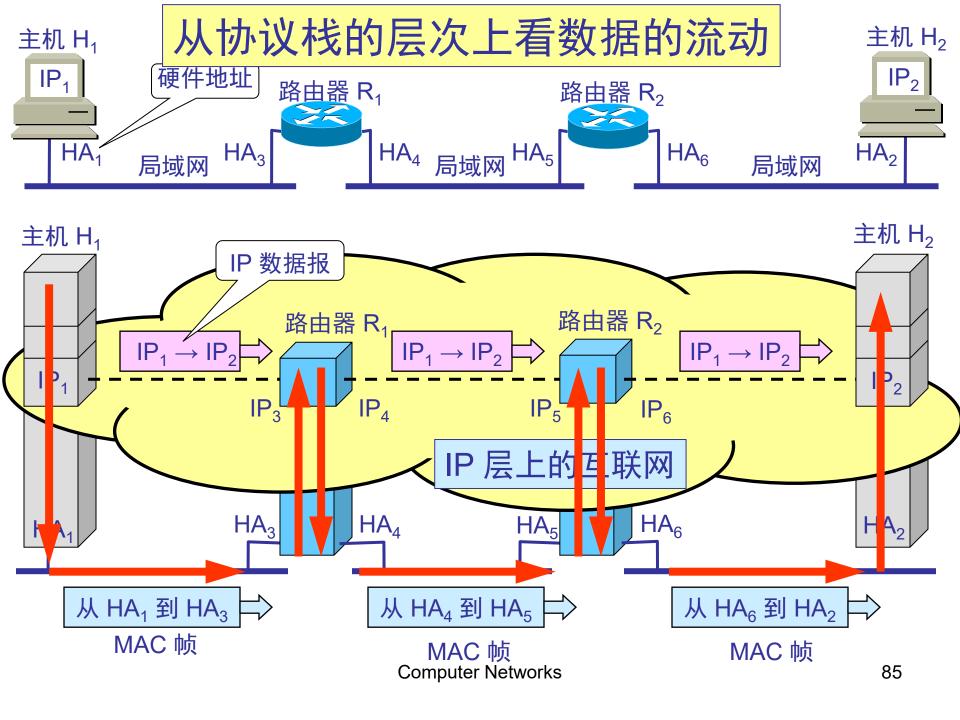
(b)不同层次、不同区间的源地址和目的地址

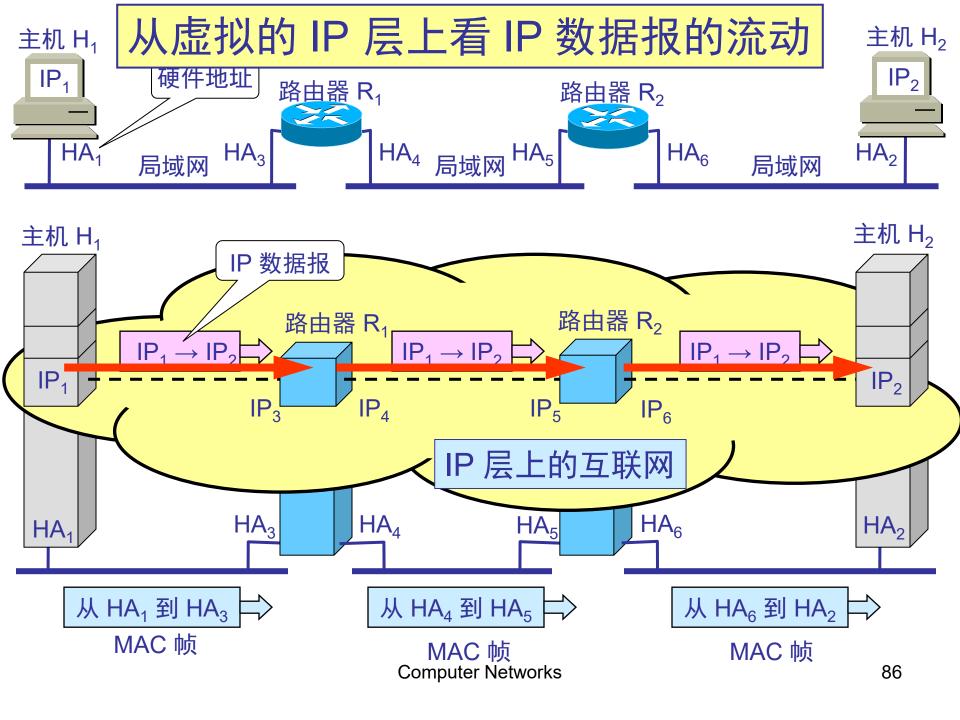
从不同层次上看 IP 地址和硬件地址

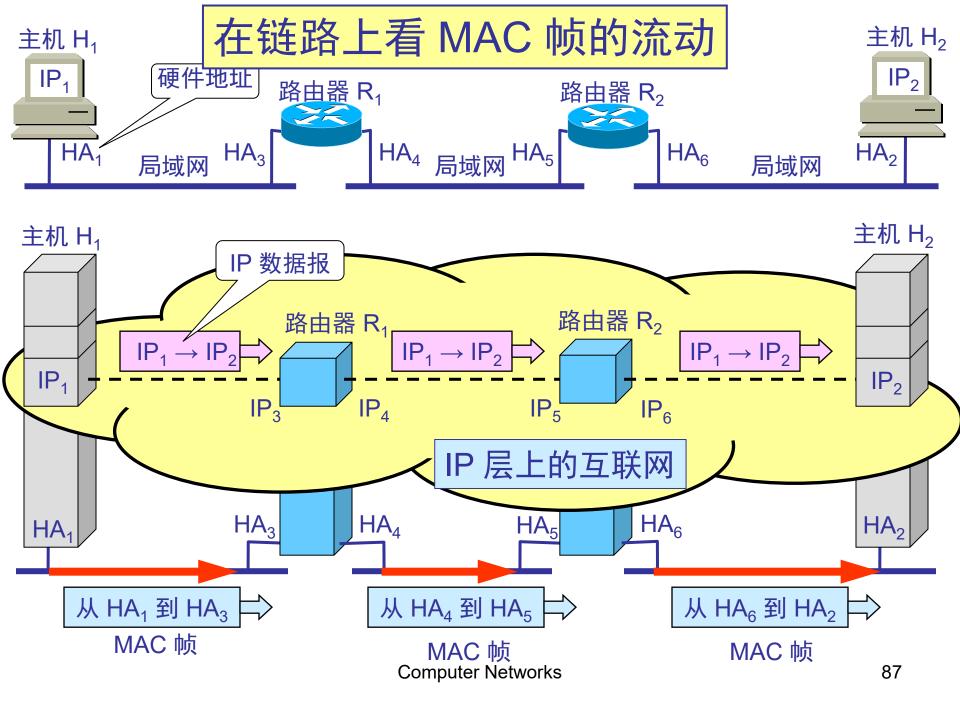


通信的路径

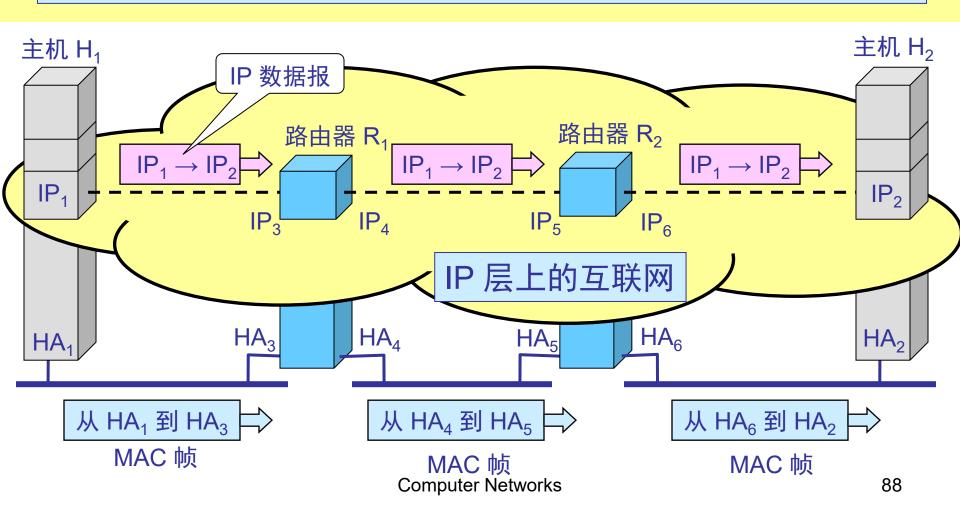
 $H_1 \rightarrow$ 经过 R_1 转发 \rightarrow 再经过 R_2 转发 $\rightarrow H_2$



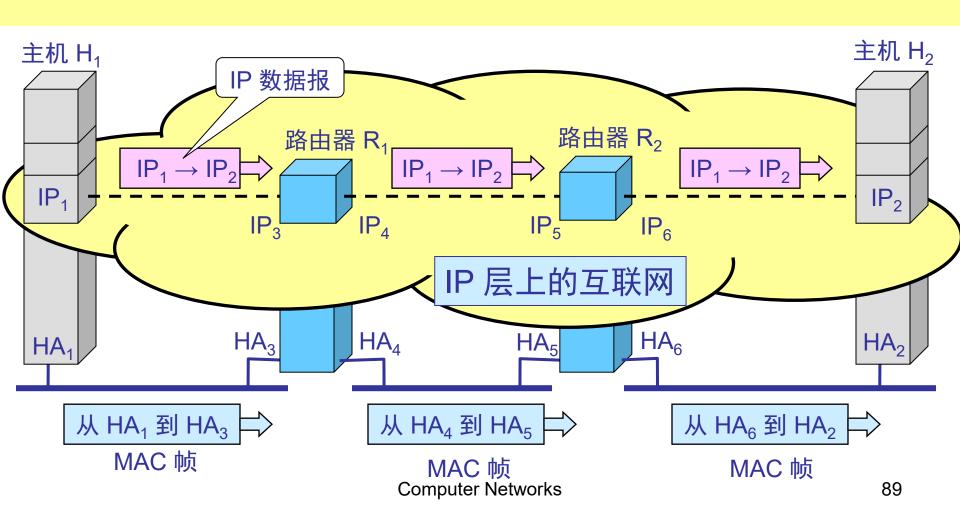




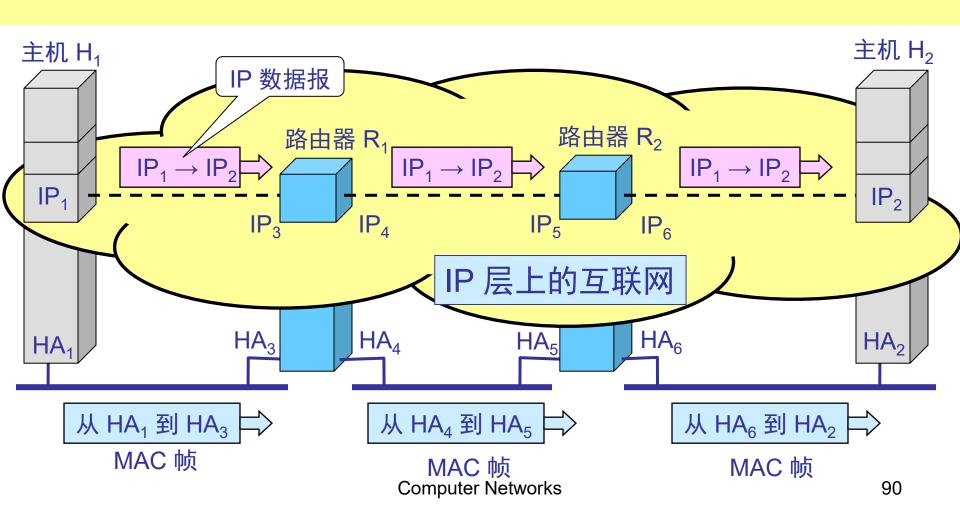
在 IP 层抽象的互联网上只能看到 IP 数据报图中的 $IP_1 \rightarrow IP_2$ 表示从源地址 IP_1 到目的地址 IP_2 两个路由器的 IP 地址并不出现在 IP 数据报的首部中



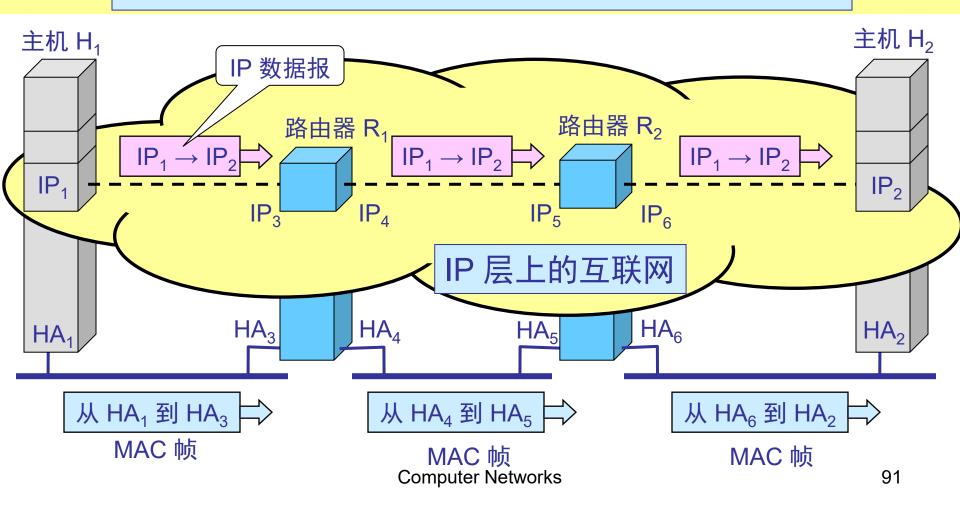
路由器只根据目的站的 IP 地址的网络号进行路由选择

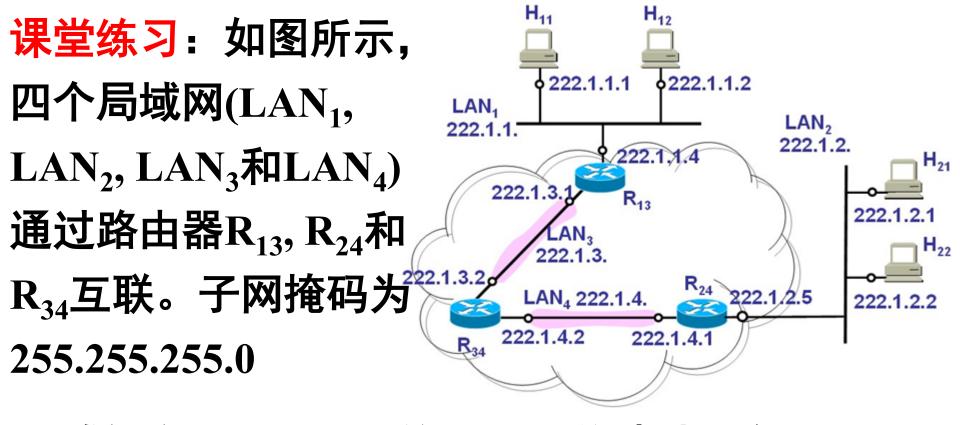


在具体的物理网络的链路层 只能看见 MAC 帧而看不见 IP 数据报

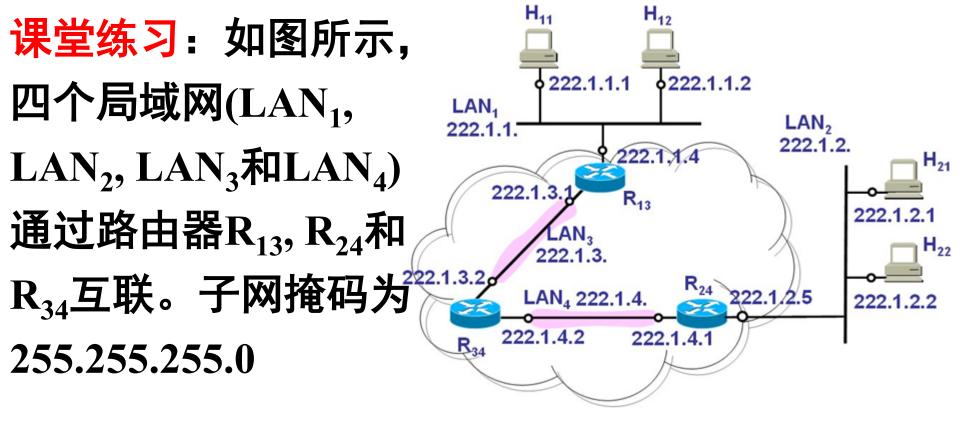


IP层抽象的互联网屏蔽了下层很复杂的细节 在抽象的网络层上讨论问题,就能够使用 统一的、抽象的 IP 地址 研究主机和主机或主机和路由器之间的通信



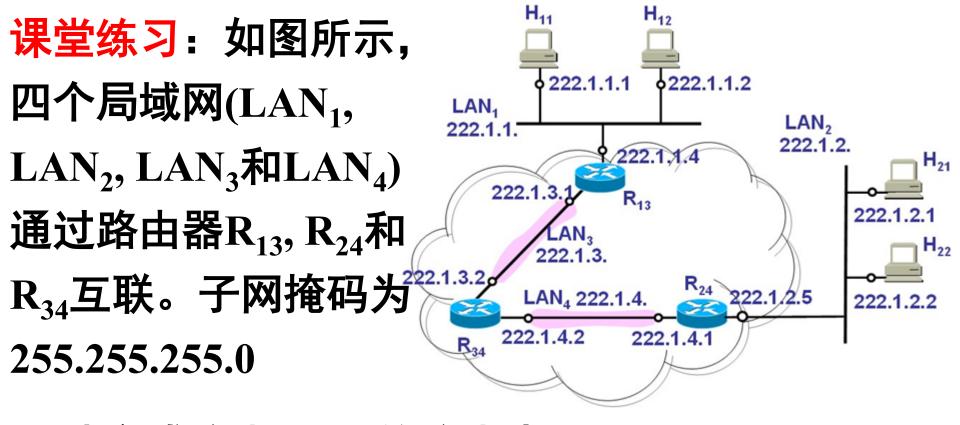


(1)试问主机H₂₁和H₂₂的默认网关地址是多少?



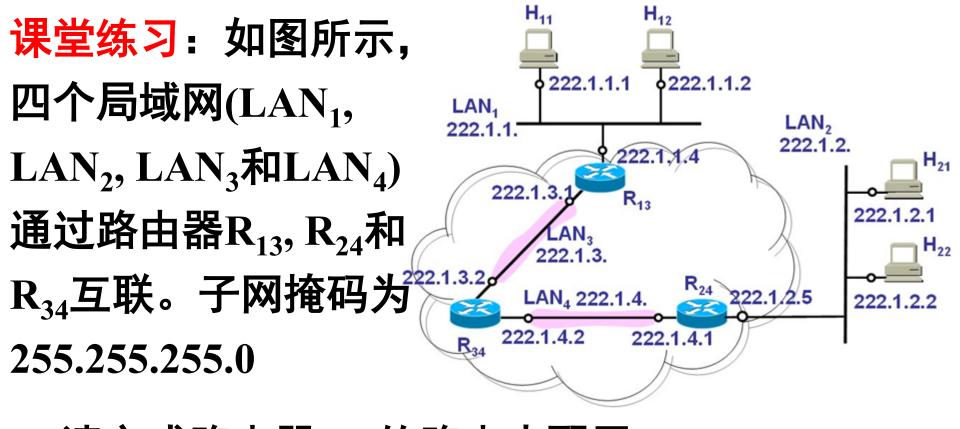
(1)试问主机H₂₁和H₂₂的默认网关地址是多少?

答: 222.1.2.5



(2)请完成路由器R34的路由表配置

目的网络	子网掩码	下一跳地址



(2)请完成路由器R₃₄的路由表配置

目的网络	子网掩码	下一跳地址
222.1.1.0	255.255.255.0	222.1.3.1
222.1.2.0	255.255.255.0	222.1.4.1

Computer Networks

Network Layer Protocols in the Internet

- IP (Internet Protocol)网际互连协议
- ICMP (Internet Control Message Protocol控制报文协议)
- ARP (Address Resolution Protocol)地址解析 协议
- RARP (Reverse Address Resolution Protocol)
 逆地址解析协议

The Transport Layer

- The service of the transport layer is to provide a virtual end-to-end "message-pipe" for applications:
 - connection-oriented
 - connectionless
 - reliable
 - unreliable

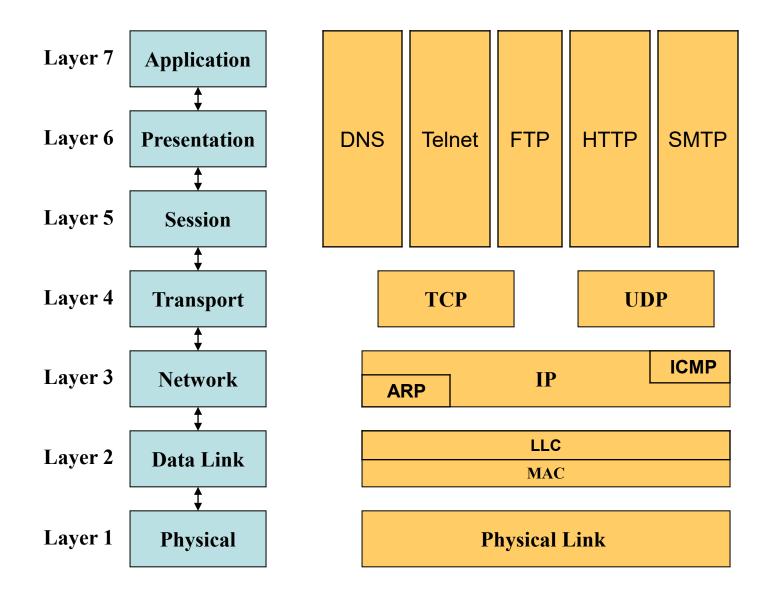
Transport Layer vs. Network Layer

- The network layer provides communication between two hosts.
- The transport layer provides communication between two processes running on different hosts.
- **Addressing: IP address vs. Port number**

Transport Layer vs. Data Link Layer

- Mat the data link layer, two adjacent nodes相 邻结点 (host-router or router-router) communicate directly via a physical link,
- Mat the transport layer, two transport entities 传输实体 within two different hosts communicate across the entire subnet.

OSI Model versus TCP/IP



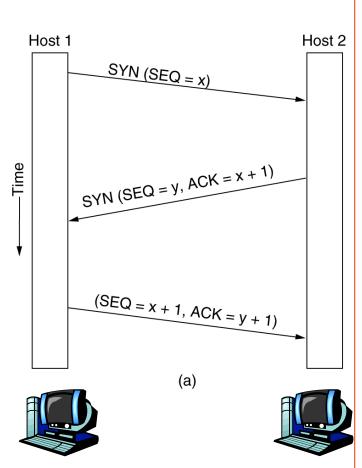
TCP (Transmission Control Protocol) 传输控制协议

- TCP provides point-to-point communi.
- TCP provides a reliable end-to-end byte stream over an unreliable IP network.
- TCP ensures that each segment is delivered correctly, only once, and in order.
- TCP is connection-oriented protocol

TCP (Transmission Control Protocol) 传输控制协议

- TCP provides full duplex communication, with Flow control and Congestion control
- ■TCP uses sliding window protocol滑动窗口协议 for flow control
- The Round-Trip Time (RTT往返时间) for TCP will change with different routes from the source host to the destination

TCP Connection Establishment



Three way handshake:

Step 1: client host sends TCP SYN segment to server with initial seq number, but no data

Step 2: server host receives SYN, replies with SYN/ACK segment

- server allocates buffers
- specifies server initial seq. no.

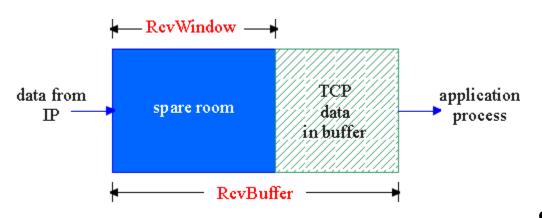
Step 3: client receives SYN/ACK, replies with ACK segment, which may contain data

Flow Control流量控制

- Flow control is a technique for preventing the sender from overwhelming the receiver with "data"流量控制是一种技术用于防止发送方用"数据"淹没接收方。
 - A receiver reserves some buffer space for storing data from a sender, while the data is being processed.
 - If the sender sends data faster than the receiver can process it, then buffer overflow will occur

Flow Control流量控制

The receive side of TCP connection has a receive buffer:



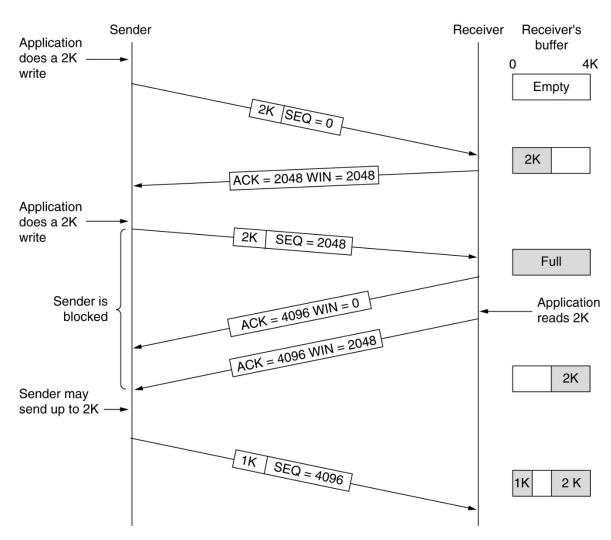
The application process may be slow at reading from the buffer

flow control
sender won't
overflow receiver's
buffer by
transmitting too
much, too fast

speed-matching service:
matching the send
rate to the receiving
app's drain rate

nputer Networks

TCP Transmission Policy传输策略



Window management in TCP.

例题1

■ 主机甲与主机乙之间已建立一个TCP连接, 双方持续有数据传输,且数据无差错与丢失。 若甲收到1个来自乙TCP报文段,该段的序号 为1913,确认序号为2046,有效载荷为100 字节,则主机甲立即发送给主机乙的TCP报 文段的序号和确认号分别是多少?

例题1,答案

解析:

- 若甲收到1个来自乙TCP报文段,该段的序号 seq=1913,确认序号ack=2046,有效载荷 为100字节,
- ■则主机甲立即发送给主机乙的TCP报文段
 - ■序号seq1=ack=2046
 - ■确认号ack1=seq+100=2013

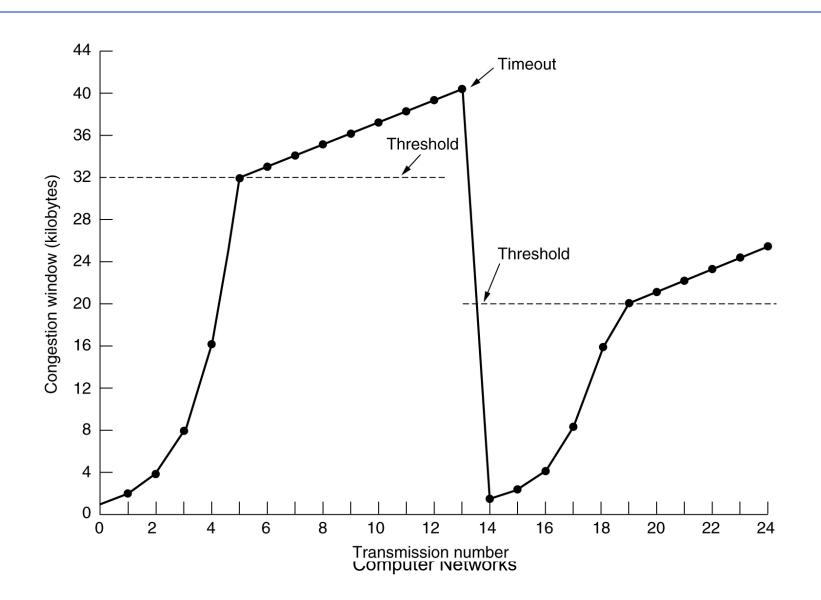
Flow Control流量控制

- Flow control at both the Data Link Layer and the transport layer
 - Stop and Wait Protocol (停止等待协议)
 - ARQ(Automatic Repeat reQuest)
 - Sliding Window Protocols(滑动窗口协议)

TCP Congestion Control 拥塞控制

- Congestion: "too many sources sending too much data too fast for network to handle"
- The purpose is to limit senders as needed to ensure load on the network is "reasonable".
- Solution :
 - Slow Start (SS)
 - Congestion Avoidance (CA)
 - Fast Retransmit (FR)

TCP Congestion Control 拥塞控制



课堂练习

如果主机A向主机B发起一个TCP连接,最大段长MSS=1KB,RTT=5ms,主机B开辟的接收缓存为64KB。则主机A从连接建立成功至发送窗口达到32KB,至少需要经过多少时间?

课堂练习

答案:从TCP连接建立好开始,主机A的发送窗口初始值为1个MSS段。

在slow start阶段按照指数规律增长:

1, 2, 4, 8, 16, 32,

经过5个RTT后,发送窗口增长到32个MSS段、即32KB。因此 5xRTT=25ms

TCP Congestion Control 拥塞控制

- Gently probe逐渐探测 network for spare capacity (SS+CA慢启动+拥塞避免)
- Drastically reduce rate on congestion冲 突发生迅速降低速率
- Retransmission on timeout超时重传
- Detecting Packet Loss/Fast Retransmit丢 包检测/快速重传
- Fast Recovery快速恢复



TCP Congestion Control 拥塞控制

Fast Retransmit + Fast Recovery

- ♥ Wait for a timeout is quite long!
- using duplicate ACKs to signal lost packet.
- Upon receipt of three duplicate ACKs, the TCP Sender retransmits the lost packet right away!

- When the TCP entity at sender always transmit data as soon as they come in from the application layer, the performance may be very poor!
- The worst case: telnet connection to an interactive editor(交互式编辑器), a TCP segment=header (20 bytes) + data payload (1 byte for one character)

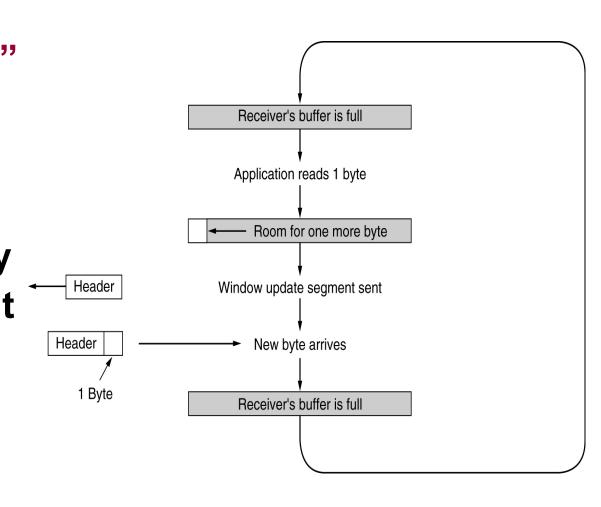
- In order to improve performance, the TCP buffers data for a moment first, then send out
- Nagle's algorithm (Nagle, 1984): when data come into the sender one byte at a time, just send the first byte and buffer all the rest until the outstanding byte is acknowledged. Then send all the buffered characters in one TCP segment and start

NAGLE, J.: "Congestion Control in TCP/IP Internetworks," Computer Communication Review, vol. 14, pp. 11-17, Oct. 1984

In 1982, Clark Receiver's buffer is full realized another problem that can Application reads 1 byte degrade TCP Room for one more byte performance, so Header Window update segment sent called Silly Window Header New byte arrives Syndrome(傻瓜窗 1 Byte Receiver's buffer is full 口综合症)

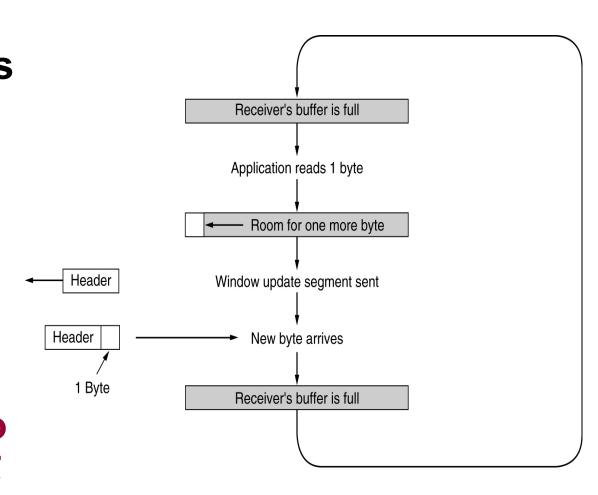
CLARK, D.D.: "Window and Acknowledgement Strategy in TCP," RFC 813, July 1982

The "Silly Window" problem occurs when data are passed to the sending TCP entity in large blocks, but an interactive application on the receiving side reads data 1 byte at a time.



Clark's solution is to prevent the receiver from sending a window update for 1 byte,

i.e. the receiver should not be too sensitive接收方不要太敏感!



- Nagle's algorithm and Clark's solution to the silly window syndrome are complementary互补的.
- The goal is for the sender not to transmit small segments and the receiver not to ask for them.

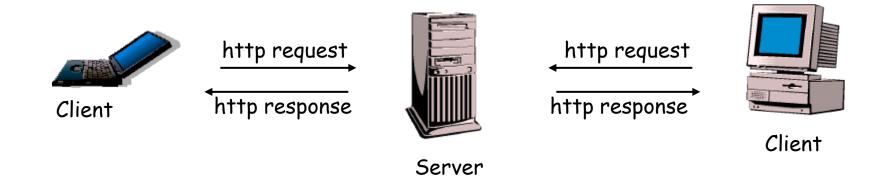
- Solution for sender side: wait until sender has enough data to transmit – "Nagle's Algorithm"
- Clark was trying to solve the problem of the receiving application sucking the data up from TCP a byte at a time.

UDP (User Datagram Protocol) 用户数据报协议

- As a connectionless transport protocol, it does not have to establish a connection to another process at the destination host before sending data.
- **UDP** has no flow control, no error control, no acknowledgements, and no mechanism to request retransmissions.
- There is no way to guarantee that the segment will reach its destination.

The Application Layer

- WWW and HTTP
- Client/Server (C/S) model



port number: 80

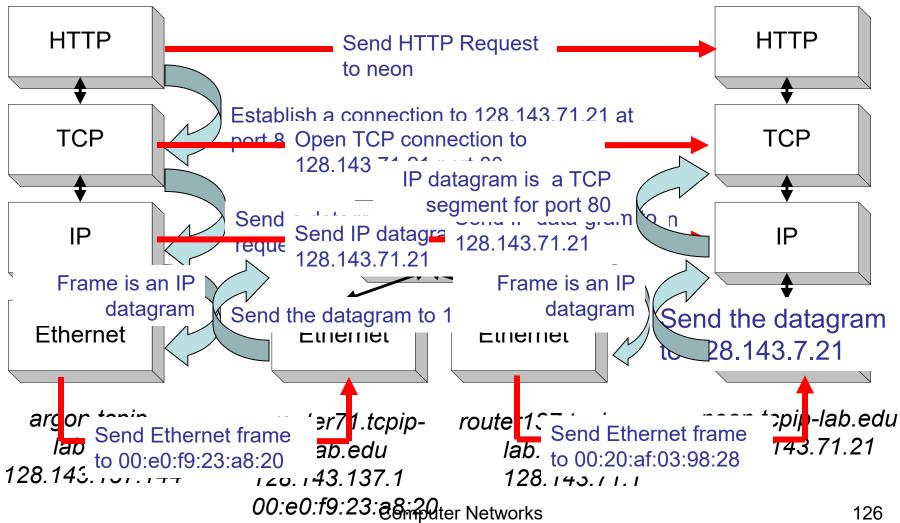
Some "Web" Terminology术语

- Web page may contain links to other pages (sometimes also called Web Objects对象)
- Object can be HTML file, JPEG image, Java applet, audio file,...
- Each object is addressable by a URL (Uniform Resource Locaters通用资源定位器*):

```
http://www.someschool.edu/someDept/pic.gif
protocol host name path name
```

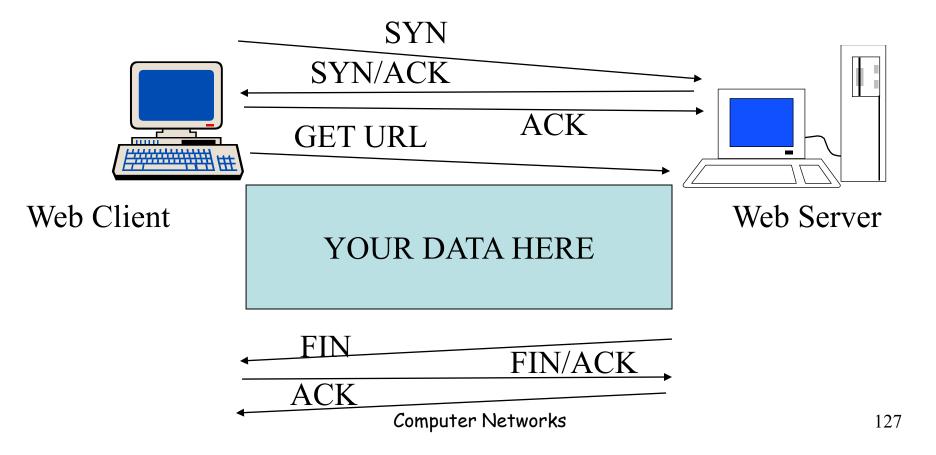
*URL: 识别Internet上的文档或资源的一种标准化"严"然。Networks

Layers in the Example



Network View: HTTP and TCP

TCP is a connection-oriented protocol



HTTP connections

Non-persistent HTTP

At most one object is sent over a TCP connection.

♦ HTTP/1.0 uses non-persistent HTTP

Persistent HTTP

- Multiple objects can be sent over single TCP connection between client and server.
- HTTP/1.1 uses persistent connections in default

Computer Networks de

Example Web Page



page.html

As you all know, the new HP book will be out in June and then there will be a new movie shortly after that...

hpface.jpg

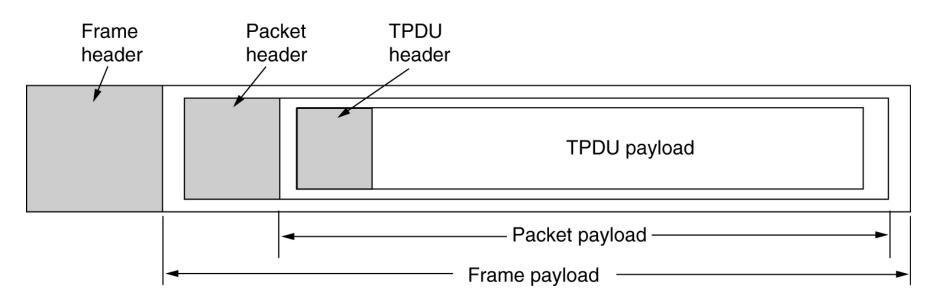
"Harry Potter and the Bathtub Ring"

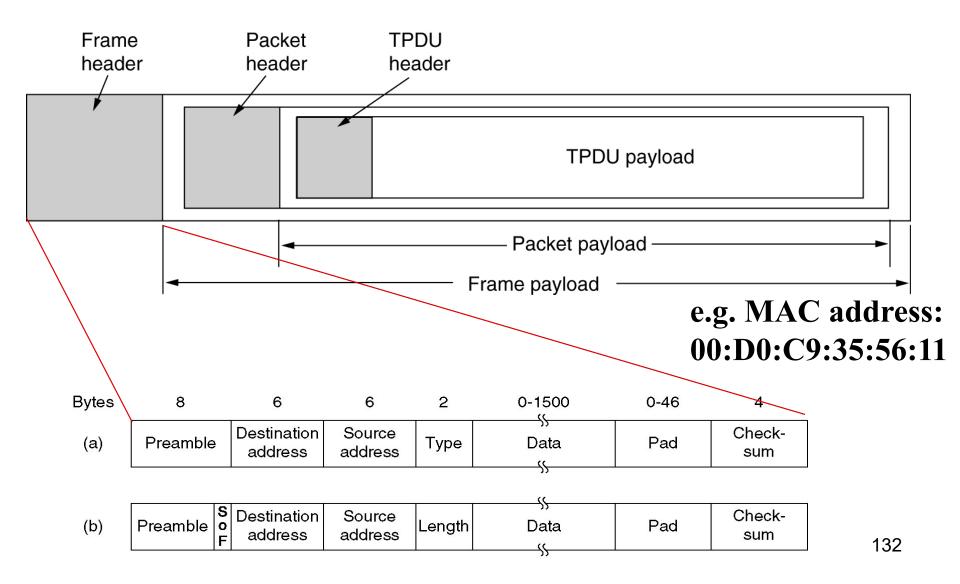


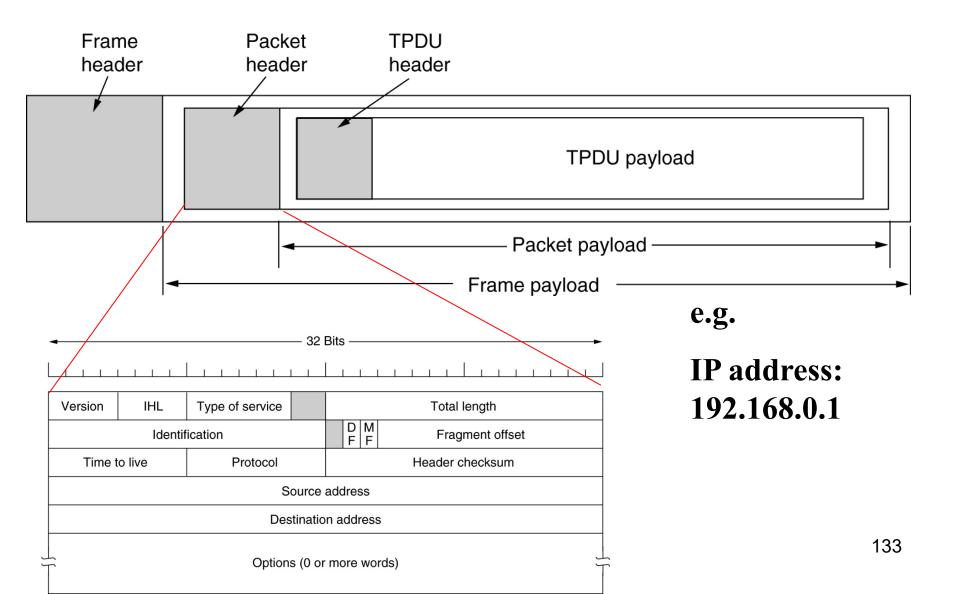
castle.gif

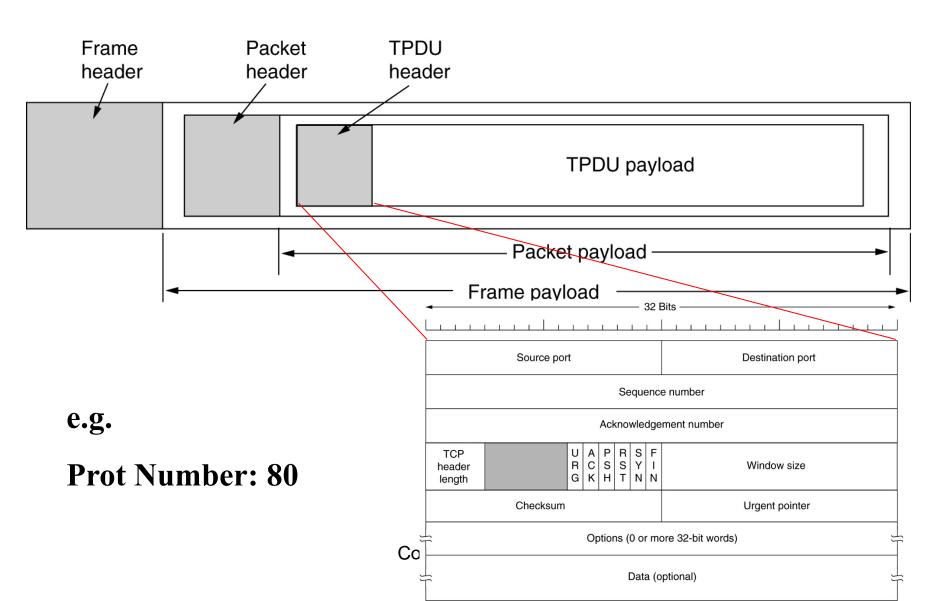
The Application Layer

- Name Servers名字服务器 is used to translate host names to IP Addresses: i.e. www.google.com → 192.168.11.11
- Electronic Mail电子邮件
 - SMTP: Simple Mail Transfer Protocol is a protocol used for sending mails
 - POP3: Post Office Protocol Version 3 is a protocol used for receiving mails
 - Port Number: 25 Computer Networks









The End!

- Good Luck with your final examination!
- Let's keep in touch!

hshen@njtech.edu.cn