Evaluation Method

- Interim and Final Report
- Attendance is not Checked, but, ...
- Questions or Comments are Mandated
  - In the quarter, questions or comments with technical content must be made at least twice during lecture (may be in Japanese)
  - Good questions and comments will be awarded with points
  - Declare your name and student ID, if you make questions or comments

Evaluation with Zoom

- questions/comments should be asked/made by oral interruption (not by chat)
  - raising hand by zoom is hard to be noticed unless dedicated chair is assigned
  - don't hesitate to interrupt my talk
    - questions/comments over chat is too easy
- name/ID and points are declared and given through chat
  - use private chat, if you don't want your ID publicly viewed

For Better Vocal Communication with Zoom

- echo cancellation of zoom is, seemingly, not very good
- it is strongly recommended to turn off speakers and use head/ear phones (should be available at 100-yen shops)

Advanced Lecture on Internet Applications
2. Transport Layer: TCP, Congestion Control, Long Fat Pipe, Multihoming

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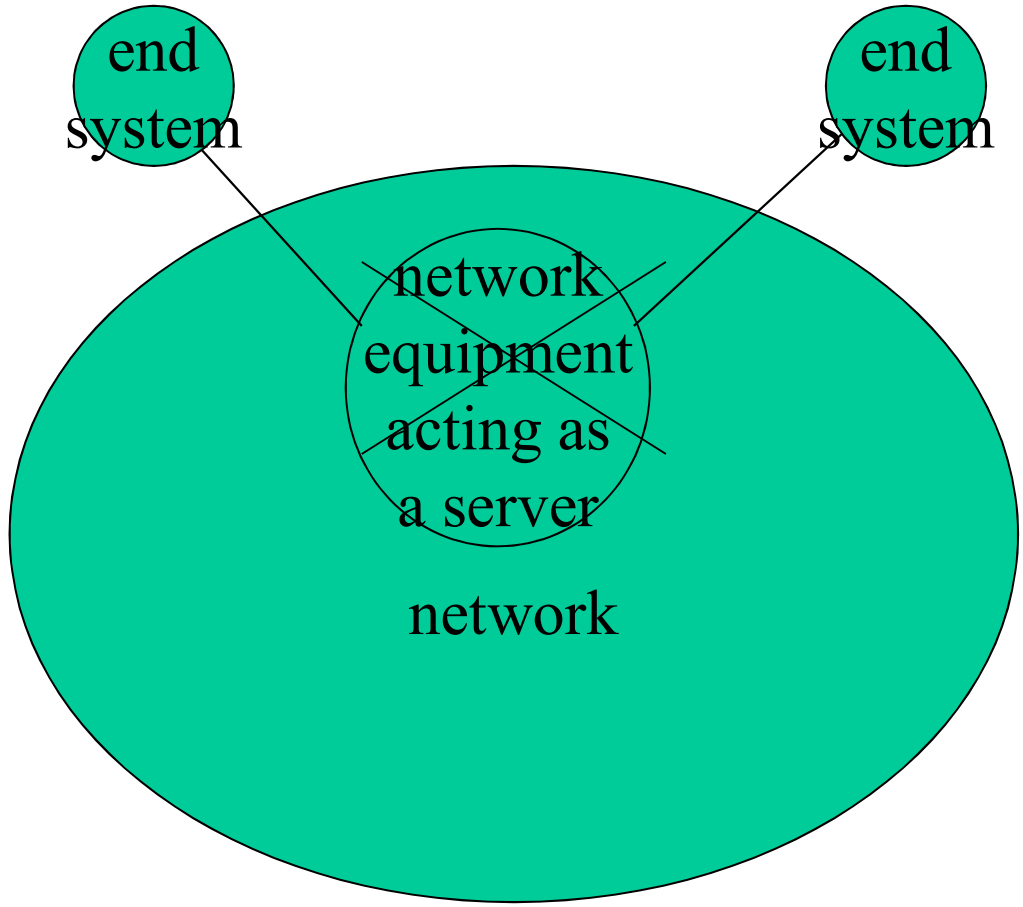
ftp://ftp.hpcl.titech.ac.jp/appli2e.ppt

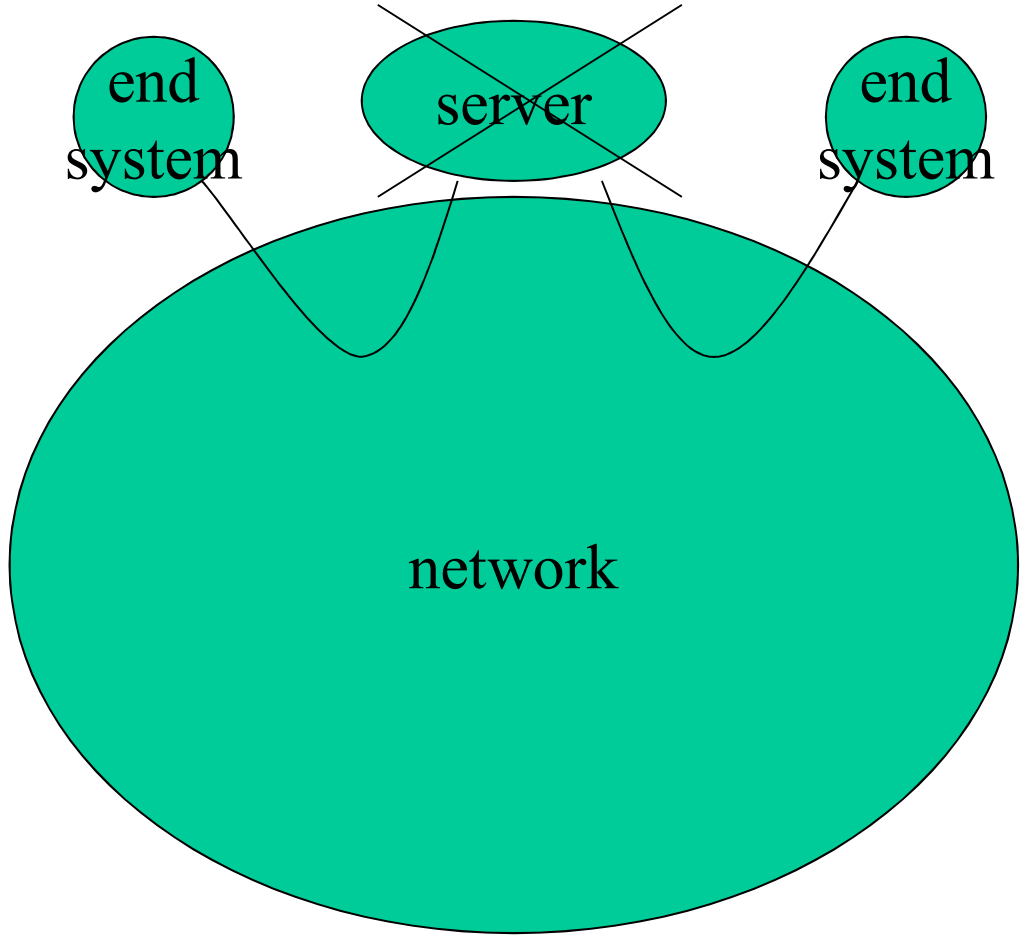
End to End Principle

The Fundamental Principle of the Internet

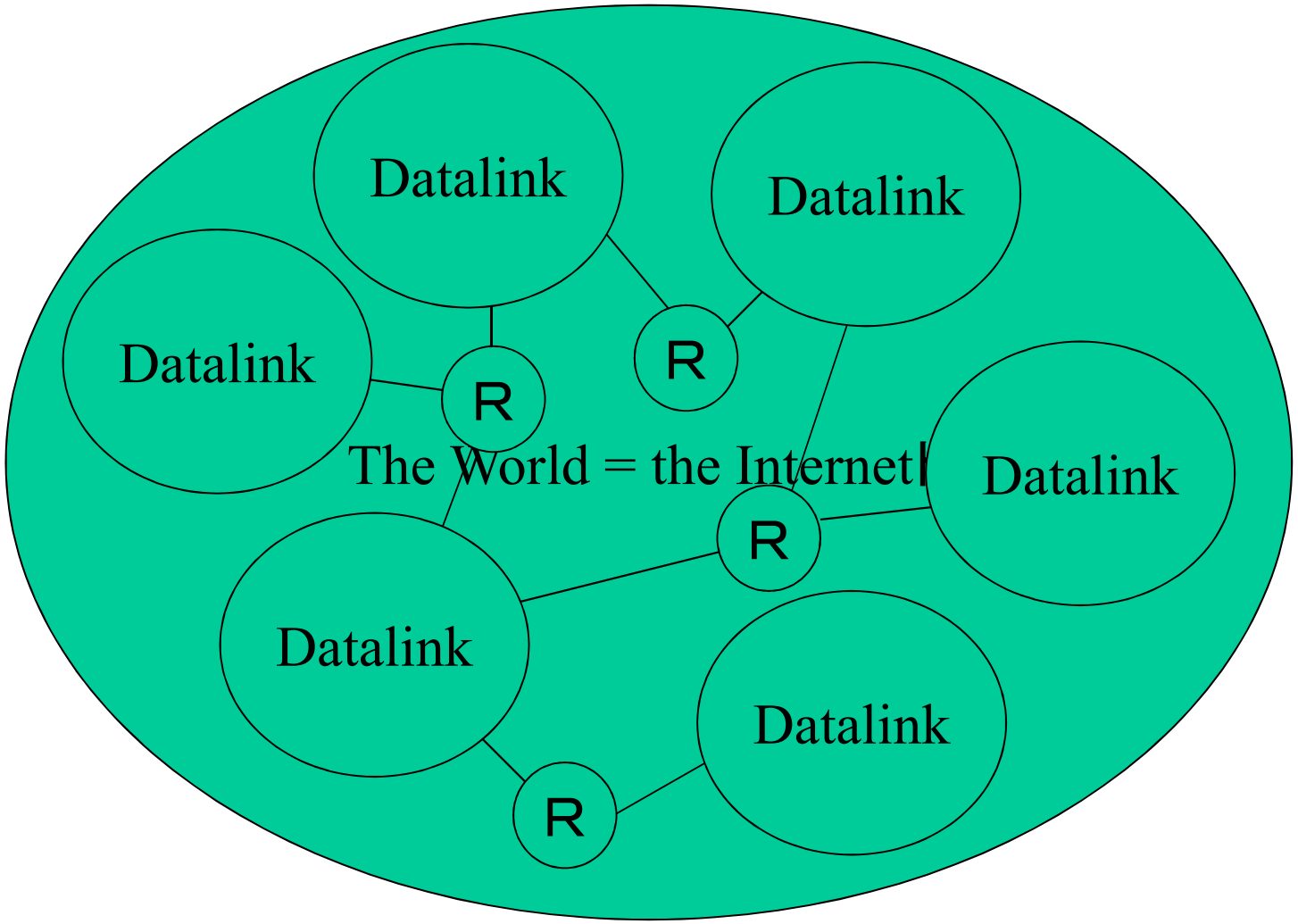
- network do as little as possible
  - only carry packets to destinations
  - no congestion control
- related end systems directly communicate
  - no intermediate intelligent servers



network do nothing

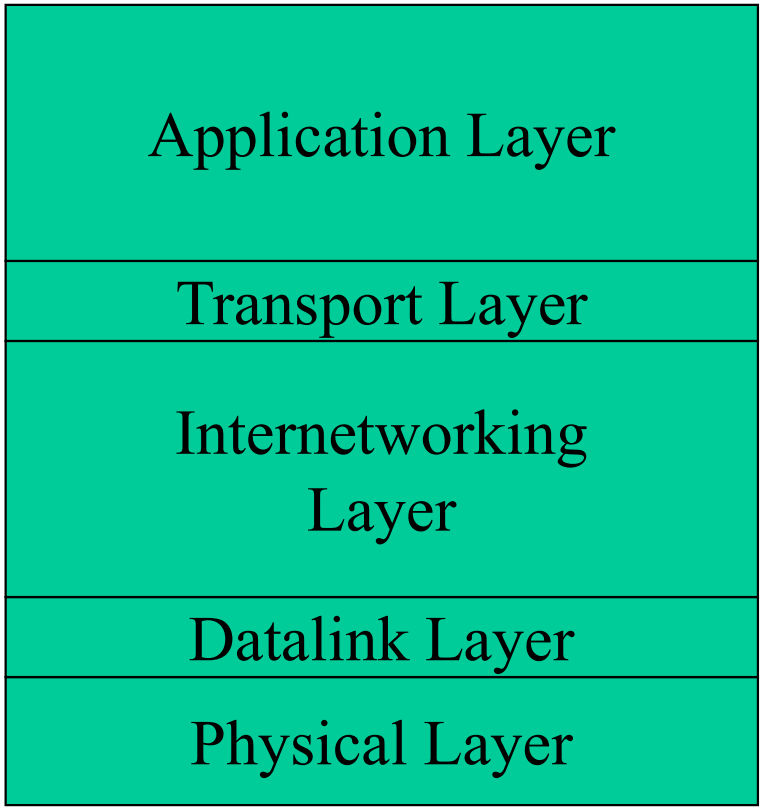


no intermediate server even outside of the network



R : Router

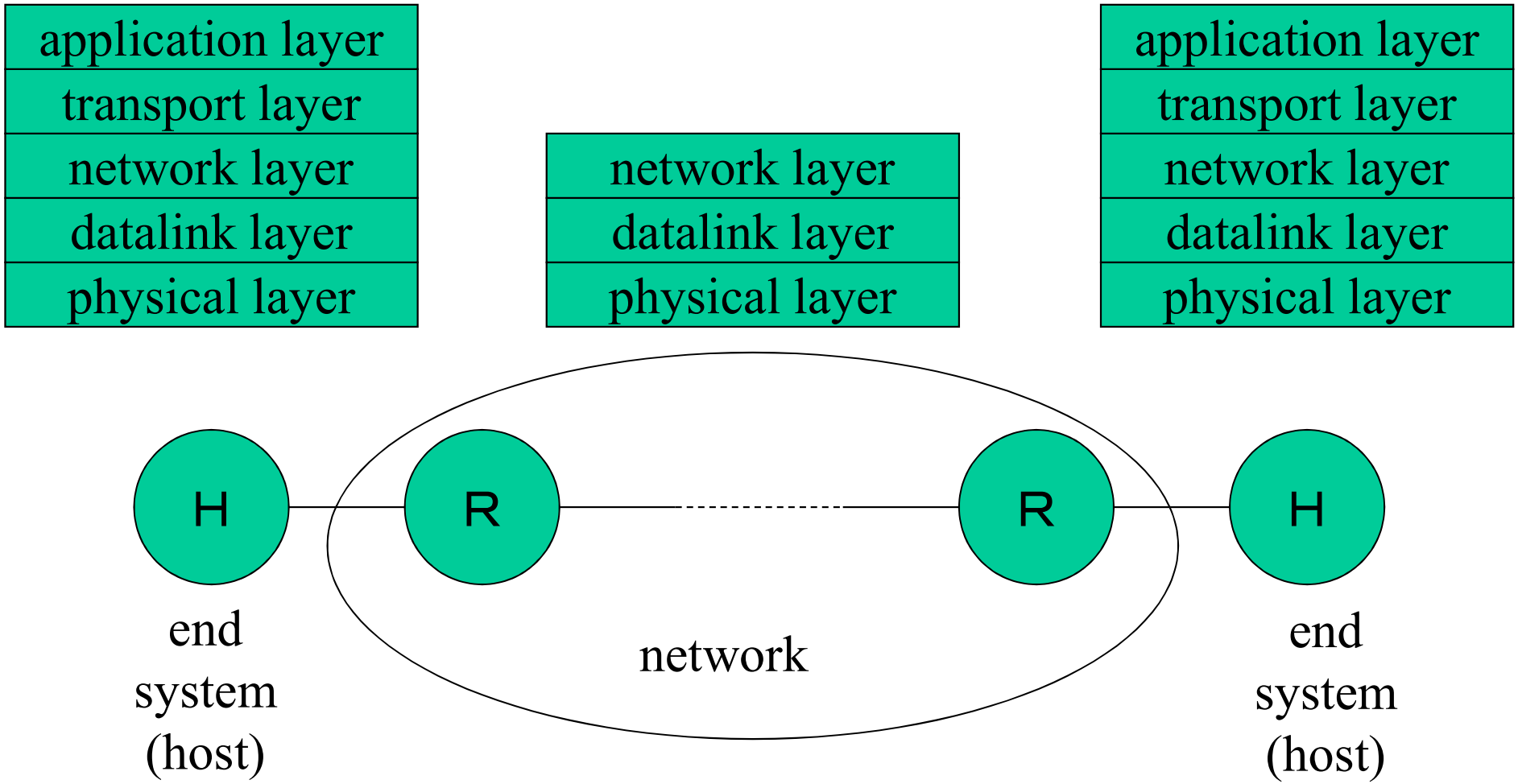
CATENET Model



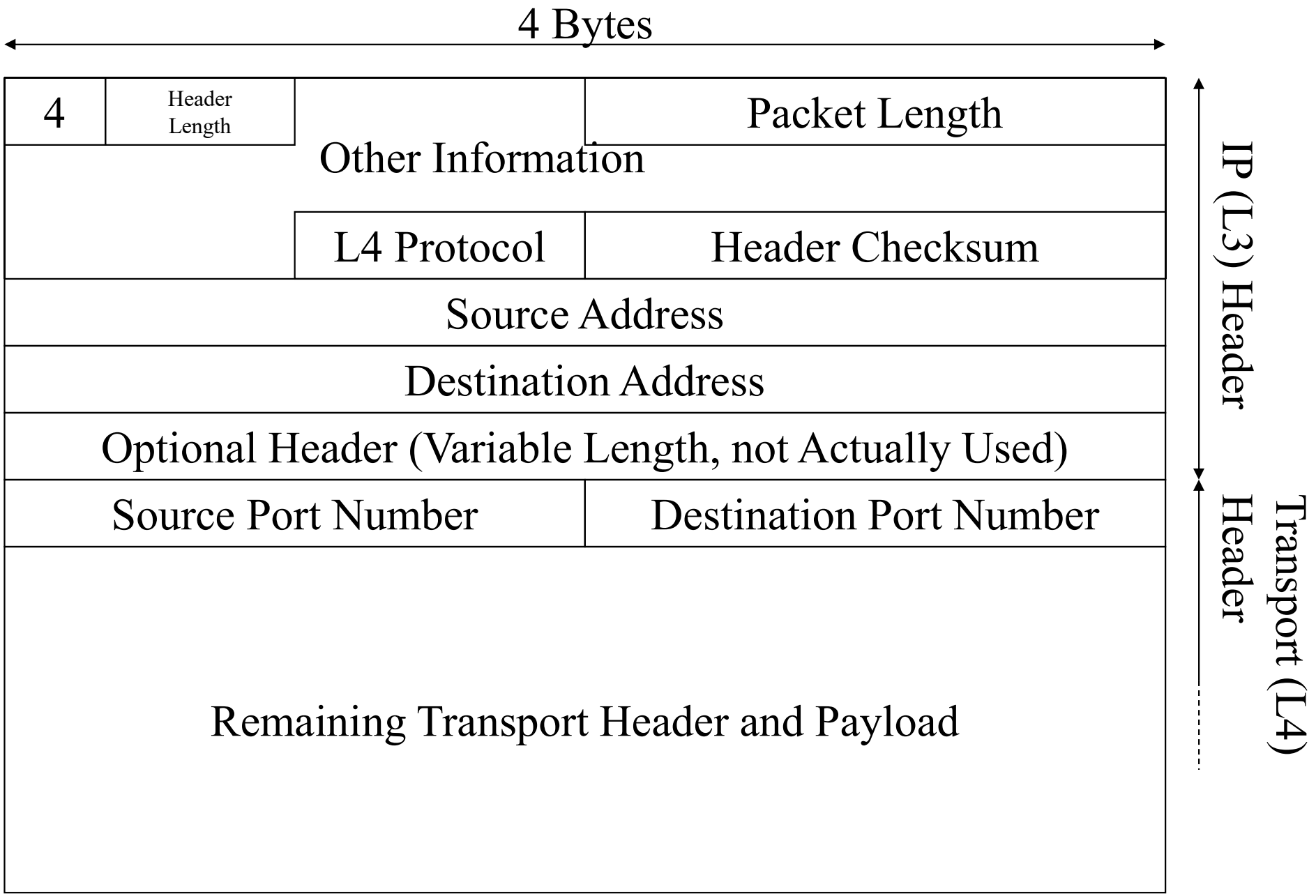
TCP, UDP

Here is the Essence of the Internet

Layering Structure of the Internet



best effort internet



Format of IPv4 Packets

Function of IP Routers

- decrement TTL and forward packet based on destination address
  - routing table is constructed in advance by routing protocols
  - no advance signaling, no BW guarantee
- with IPv4, may divide packets for datalinks with small MTU (fragmentation)
- IPv4 (rfc791) address is 32bit long
  - transition to IPv6 with 128bit address planned

TCP and UDP

- TCP (rfc793)
  - Transmission Control Protocol
  - retransmit when data error or drop is detected
  - adjust transmission rate
- UDP (rfc768)
  - User Datagram Protocol
  - do nothing (let applications do something)
    - nothing except for delivery to applications

Phone Network vs the Internet

- which is more error prone?

Phone Network

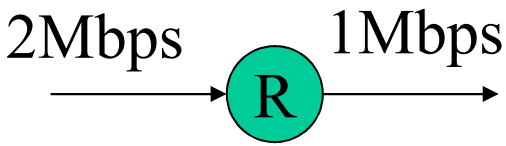
- network for voice transmission (conversation)
- guarantee bandwidth
- minimize delay
  - less than 0.1s of delay desirable for conversations
  - no time for retransmission of lost data (almost impossible with analog circuit)
- error or loss of data causes noise
  - if not very frequent, not a serious problem for conversation

Internet

- network for communication between computers
- a bit of error is often fatal
  - character code, program
  - detection of error or loss required
  - reliable communication over TCP is extensively used
- UDP may be used for voice
- light weight protocols (DNS, tftp, etc.) also use UDP

Reason of Packet Drop

- packet is lost upon transmission error
  - not so common
- routers must drop packet if buffer is full
  - primary reason of packet drop in the Internet



Congestion Control in Phone Network

- no control necessary because bandwidth is reserved?
- connections fail during bandwidth reservation
  - you will here busy tone
    - not your destination but the network is busy

TCP

- connection oriented
- principle is simple
- provisions for various exceptions are complex
- theory and implementation for rate control to avoid congestions is also complex
  - taken care of end systems only

TCP and the Internet

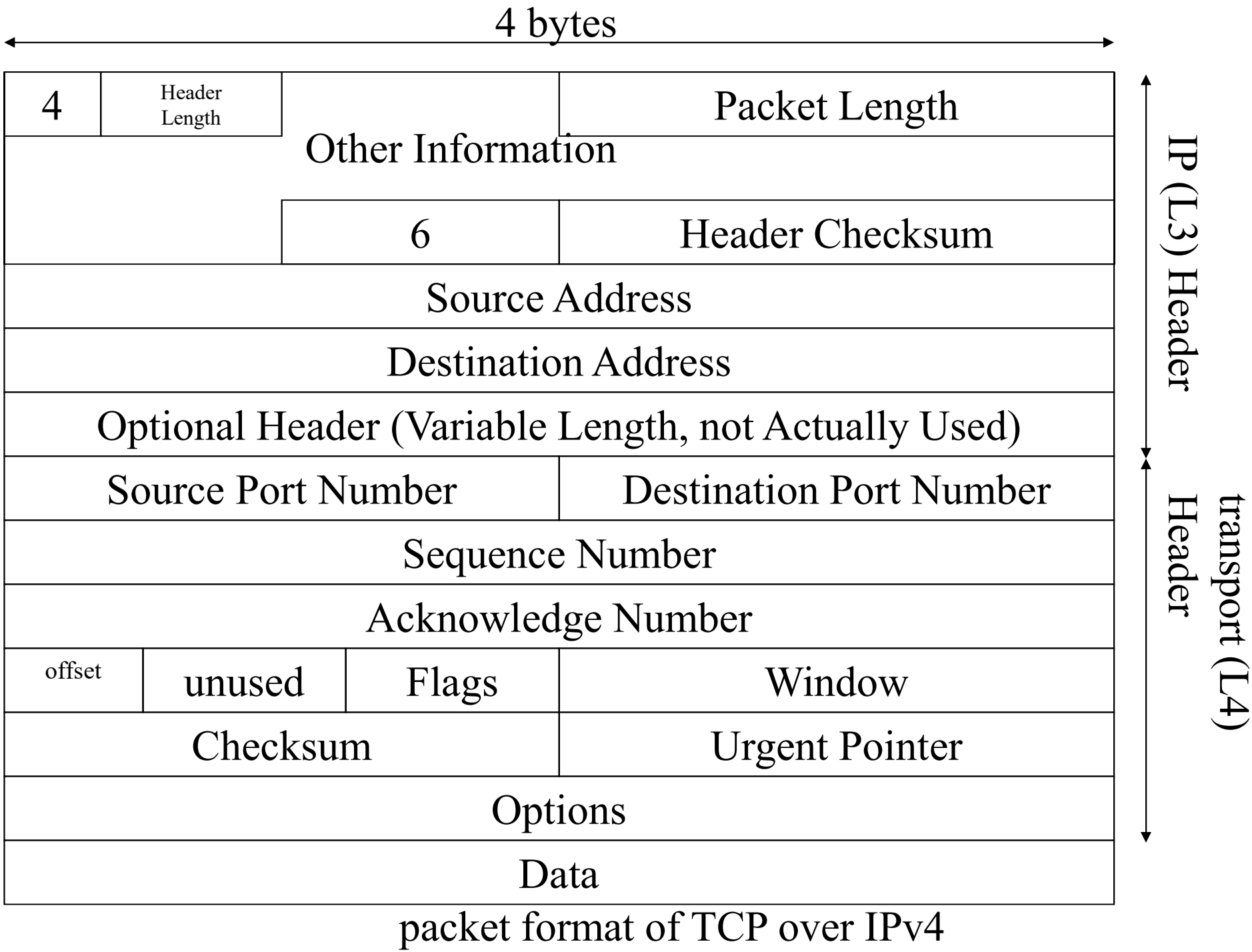
- TCP is the most important transport protocol of the Internet
  - used by almost all applications
  - congestion avoidance of TCP makes the Internet barely operational
    - will collapse?
- TCP/IP protocol suites
  - acronym of set of protocols related to the Internet

Basic Operation of TCP

- initiate connection
  - share sequence numbers to count bytes
- transmit data
  - transmit data within window size
- acknowledge reception of data
  - acknowledge data by sequence number
- retransmit data
  - retransmit unacknowledged data

Reliability and E2E Principle

- there is no error free network
  - retransmission is necessary to recover lost information
  - buffering within network for retransmission
    - fails if the buffer is involved in the error
    - can not adopt to route changes
    - maybe useful if error rate is high "as a performance enhancement"
  - end must hold
    - do not have to make network so much reliable



packet format of TCP over IPv4

Source Port Number, Destination Port Number, Checksum

- same as UDP
  - except for the location of checksum field
- connection is identified by a quadruple of (source address, source port number, destination address, destination port number)
- checksum is mandated

Sequence Number

- sequence number of data (in byte) assigned by source
- initial value is randomly determined
- wraps around to 0 after 0xffffffff

Acknowledge Number

- sequence number reception of which is acknowledged by destination

Offset

- TCP header length (32bit wise)

Flags

- URG: Urgent Pointer field significant
- ACK: Acknowledgment field significant
- PSH: Push Function
- RST: Reset the connection
- SYN: Synchronize sequence numbers
- FIN: No more data from sender

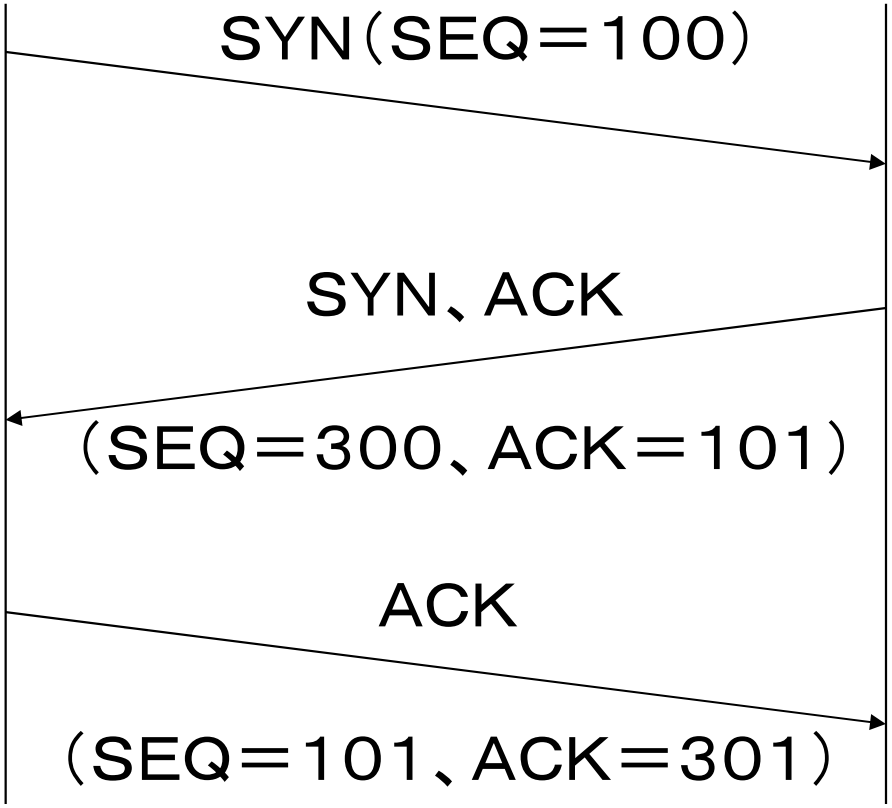
Window

- amount of data source can send before acknowledgement
  - amount of buffer at receiver
- to send data in high speed, window must be large
  - speed=(window size)/(round trip time)
- to avoid congestion, source further limits window size

Urgent Pointer

- location of urgent data in data

Connection Establishment of TCP
3-way Hand Shaking



Congestion Control

- BW is not managed in the Internet
- if everyone send packet at will, large amount of packet loss may occur
- everyone will be happy if packets are sent at rate a little below link BW
- though merely gentlemen's agreement
  - combined with TCP, widely spread
  - can not break the agreement unless both sides cooperate

Practice of Congestion Control by TCP

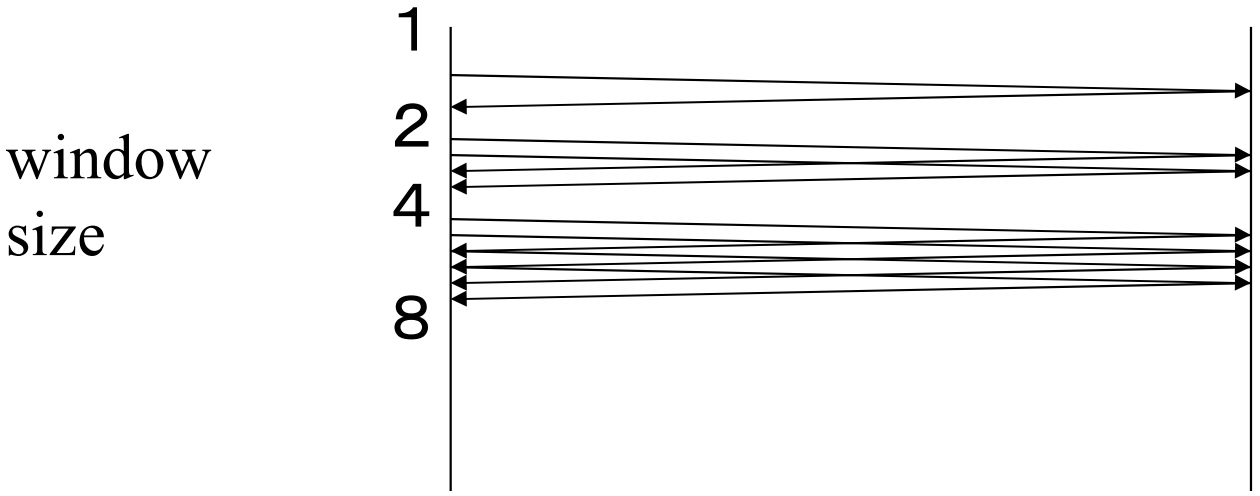
- control TCP rate according to congestion situation of the Internet
  - if congested, reduce transmitter's window size
- what is congestion?
  - packet drop
- packet drop is detected by timeout or acknowledge number not increasing

Congestion Control and E2E Principle

- action for congestion by TCP
  - upon congestion, routers tries buffering and, if buffer is full, just drop packets
    - large buffer is harmful
  - end systems estimate congestion situation in the network and perform complex control (rfc2001)

Slow Start

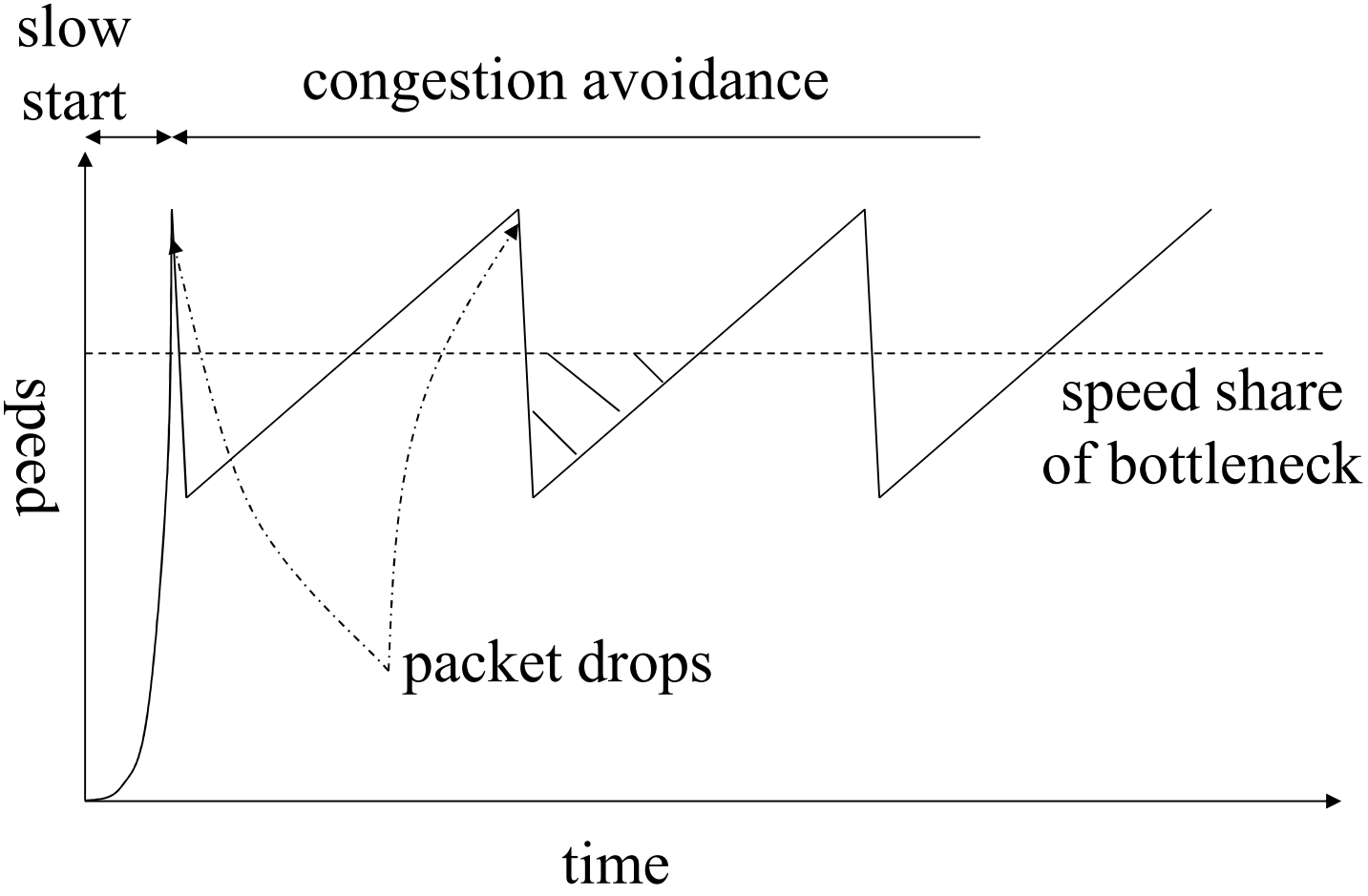
- initial window size is minimum
- if ACK is smoothly replied, increase window size with a fixed amount
  - speed increase exponentially



Congestion Avoidance

- if packet drop is detected
  - window size is halved
- then, window size is increased inversely proportional to the current window size
  - speed increases linearly

Traffic Variation of TCP



Path MTU Discovery

- MTU
  - Maximum Transfer Unit
  - maximum packet size carried by datalink
    - different datalink by datalink (1500B for Ethernet)
- path MTU
  - minimum MTU of path between ends
- path MTU discovery (PMTUD)
  - estimate PMTU

Why PMTUD Necessary?

- with IPv4, packets larger than MTU is divided (fragmented) by intermediate routers
  - heavy weight processing
    - not allowed with IPv6
- larger packet size means less overhead by headers (BW and processing)
- packets of PMTU size is most efficient

Reality of PMTUD

- set “don’t fragment” bit in IPv4 header
  - ICMP error is returned if MTU is exceeded
    - ICMP packet contains MTU of next hop (rfc1191)
- periodically try to send larger packets
  - because route changes may mean MTU changes
- not very practical
  - ICMP packets are often filtered
  - not usable with multicast
  - periodic ICMP error generation increases load

Extension to Congestion Control

- by modifying routers
- RED
  - Random Early Drop
- ECN
  - Explicit Congestion Notification

RED

- tail drop
  - drop packets if buffer is full
- random early drop
  - drop packet with low probability if buffer is occupied to some extent
- can initiate TCP congestion control in early stage

ECN

- mark packets if packet drop is likely to occur
  - use 2 bits of ToS field of IP header
- questionable effect
  - RTT amount of delay to source, anyway
    - not very different from packet drop
    - though packets are not dropped

Collapse of the Internet (game theoretic instability)

- lost packet may be restored by ECC
  - FEC (Forward Error Correction)
- under severe congestion
  - using strong FEC makes the user happy
    - though the FEC increases traffic
  - if all use the FEC, congestion becomes worse
    - beyond correction capability of FEC
- BW guarantee in network inevitable?

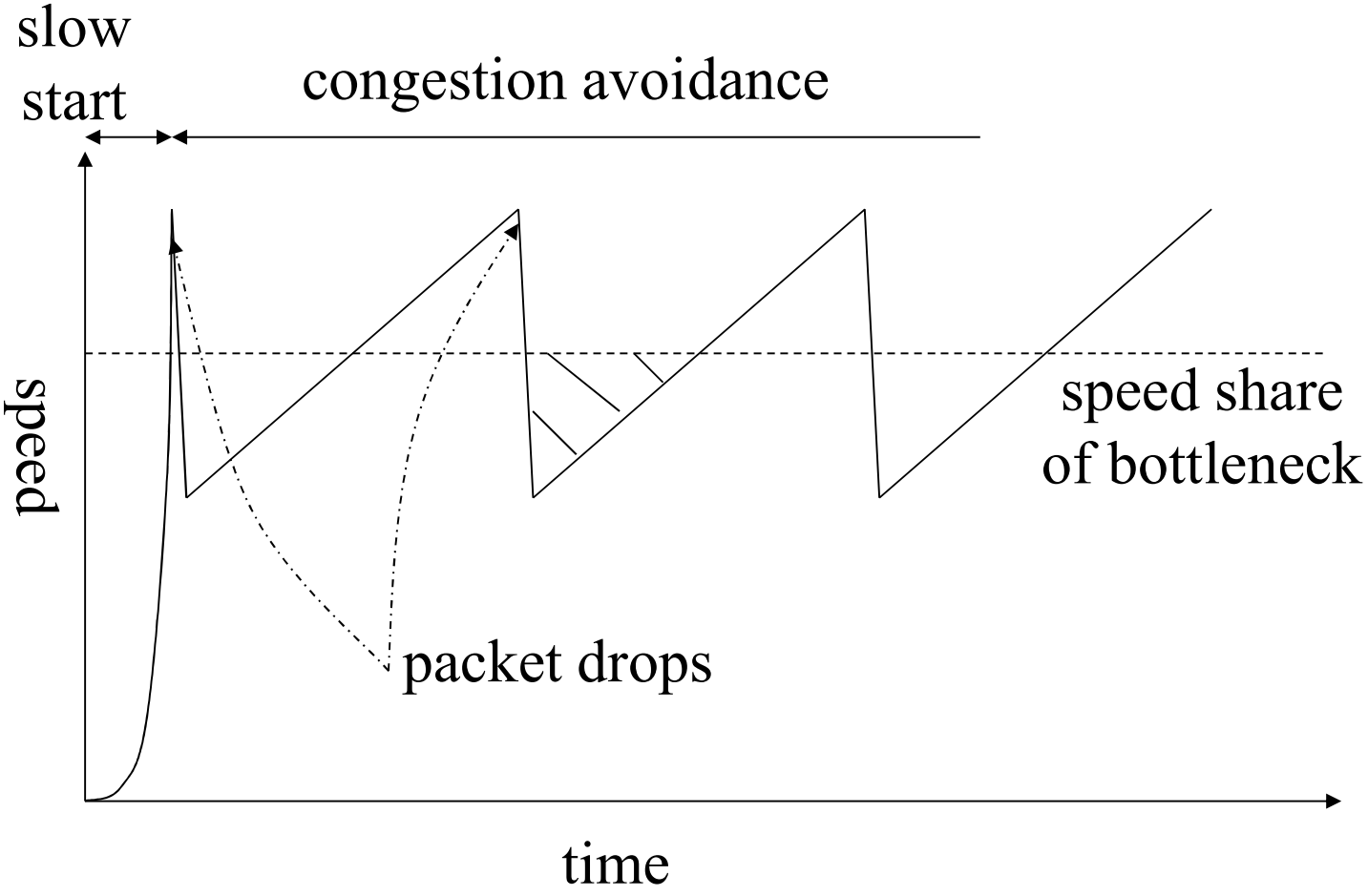
Long Fat Pipe

- TCP performance is limited by RTT
  - speed=(window size)/RTT
- RTT is large for long distance link
  - performance degrages
    - 5Mbps with 0.1s RTT and 64kB window
- packet drop can be disastrous
  - all data within RTT must be resent
- various workaround such as SACK (rfc2018)
  - ultimate solution is BW guarantee

TCP and Router Buffer

- CA makes traffic variation like saw tooth
- without buffer, link speed can not be used up
  - (link(?) delay)*(link BW) of buffer necessary
- backbone routers needs large buffer?
  - backbone is fast
  - backbone is long

Traffic Variation of TCP



TCP and Backbone Router Buffer

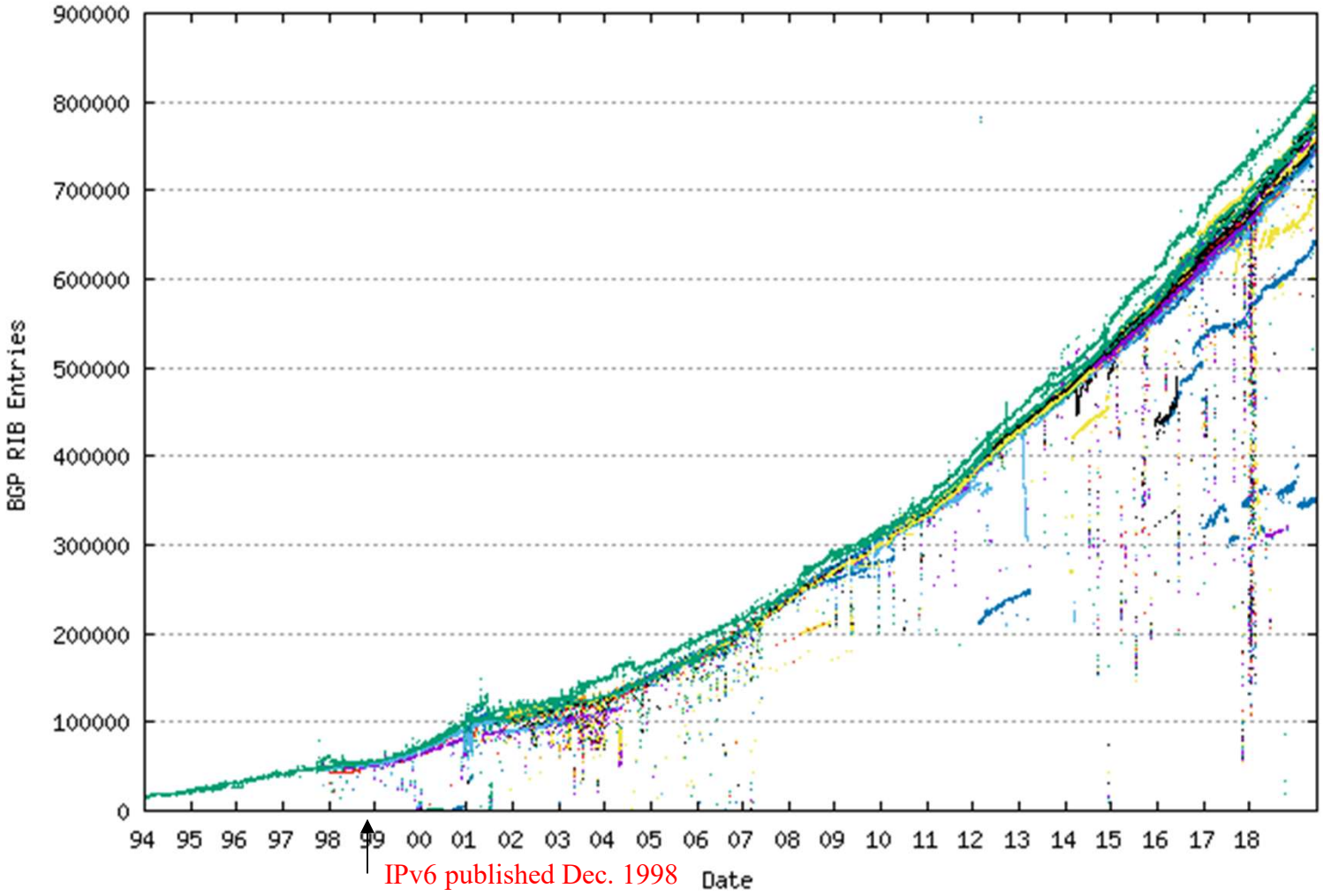
- backbone routers need huge buffer?
  - at backbone, variations of many (*N*) TCP are averaged (if variations are independent)
  - variation is 1/sqrt(*N*)
    - need 1/sqrt(*N*) less buffer?
    - if several times of 1/sqrt(*N*) of link speed is sacrificed
      - traffic rarely to exceed link capacity (if poisson)
    - buffering of several tens of packets to absorb short term variation is enough
      - » optical packet router is plausible

Function of IP Routers

- decrement TTL and forward packet based on destination address
  - routing table is constructed in advance by routing protocols
  - no advance signaling, no BW guarantee
- with IPv4, may divide packets for datalinks with small MTU (fragmentation)

IPv4 Routing Table Size

http://bgp.potaroo.net/

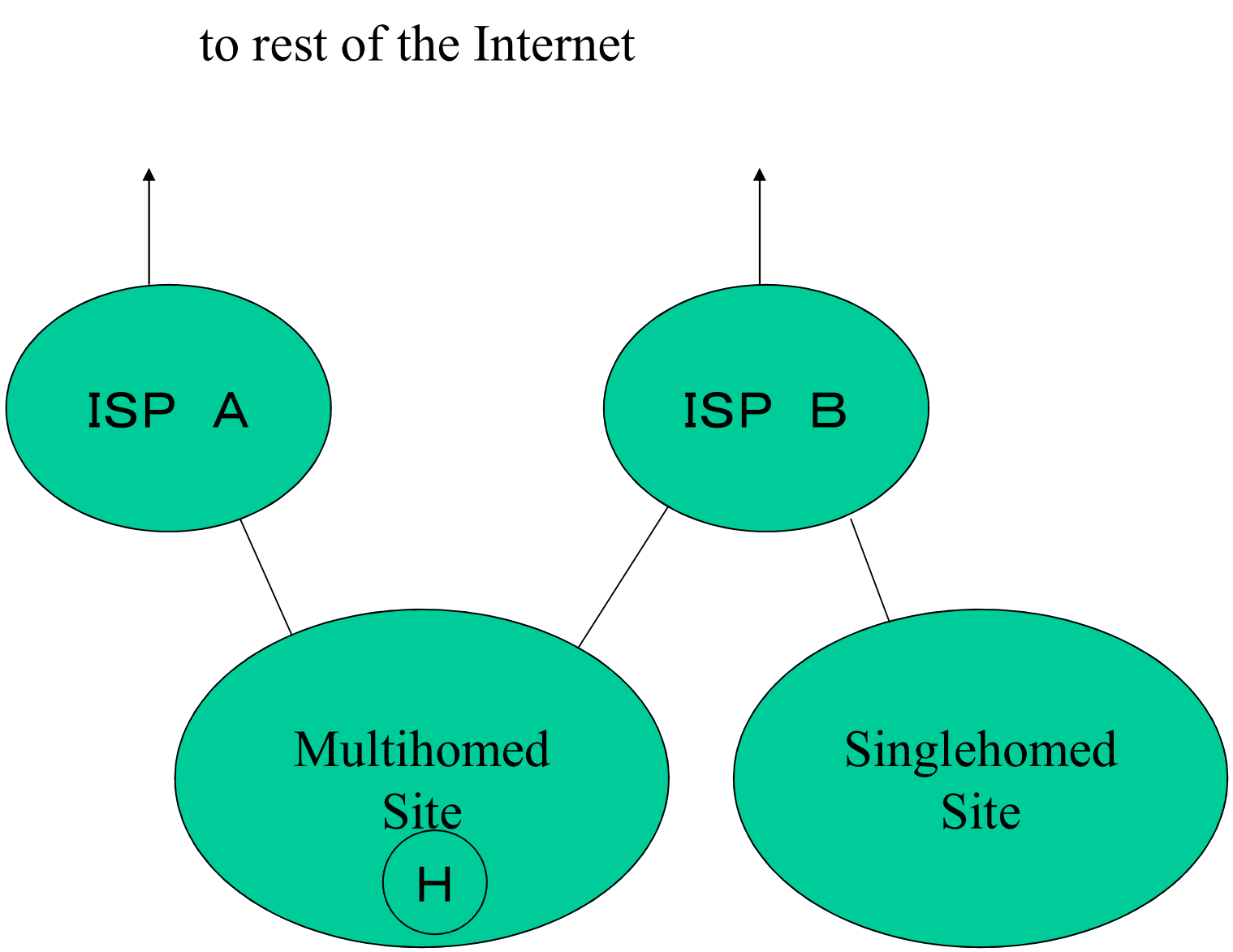


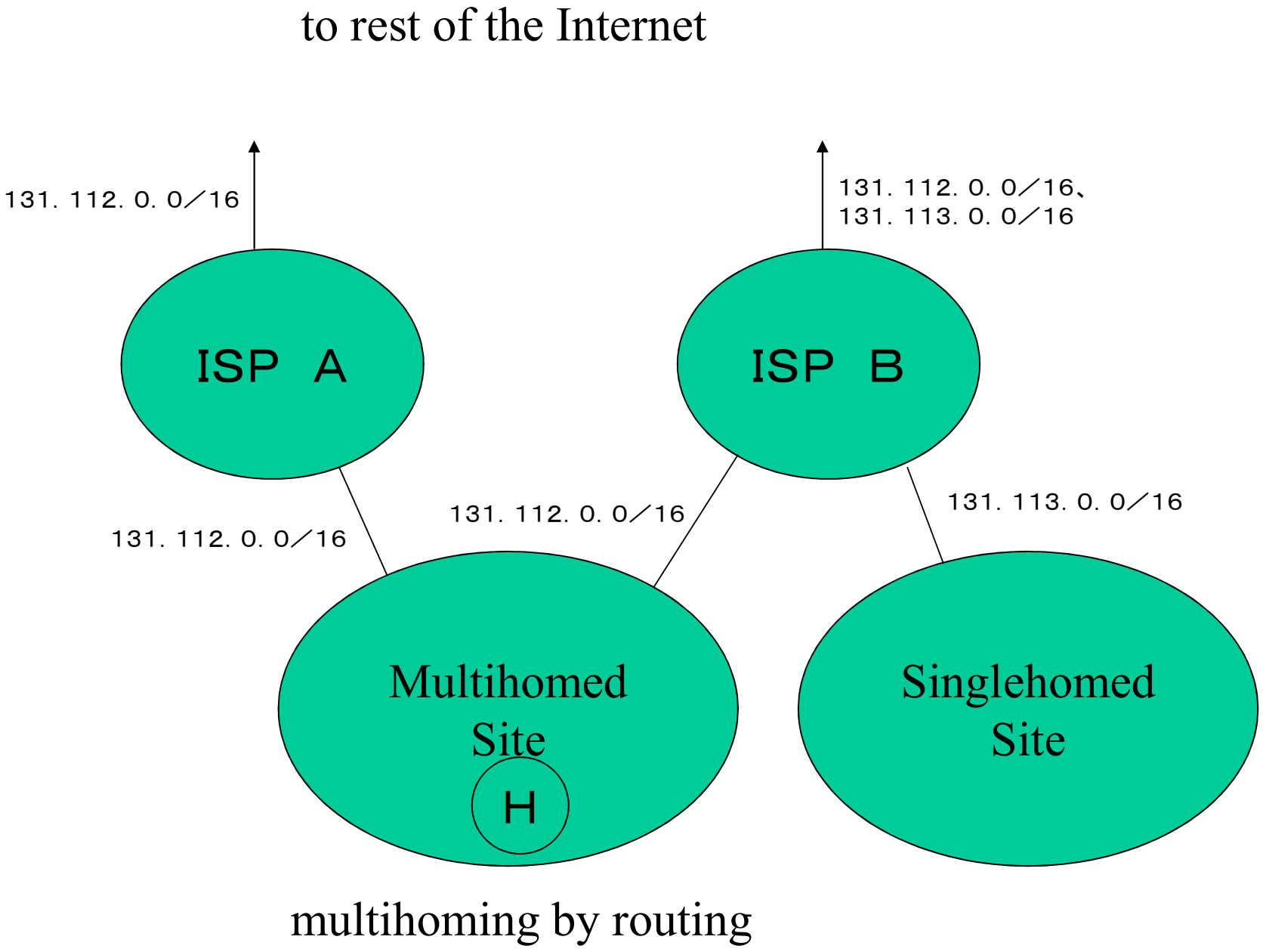
Cases When Route Aggregation Impossible

- aggregation possible, if route is shared by addresses sharing a pattern
- route not by destination address only
  - QoS routing depends on required QoS
- destination address not designate location
  - multicast address designate set of locations
- random IP addresses within a region
  - initial allocations for IPv4
  - multihoming by routing

Multihoming

- have multiple upstream ISPs
  - safe even if some ISPs fail
- necessary for reliable service
  - police, ISP, banks, large corporations etc.
- multihoming by routing send single address range to multiple pathes
  - let routing protocol of the network choose the better (or available) one


to rest of the Internet
ISP A
ISP B
Multihomed
Site
H
Singlehomed
Site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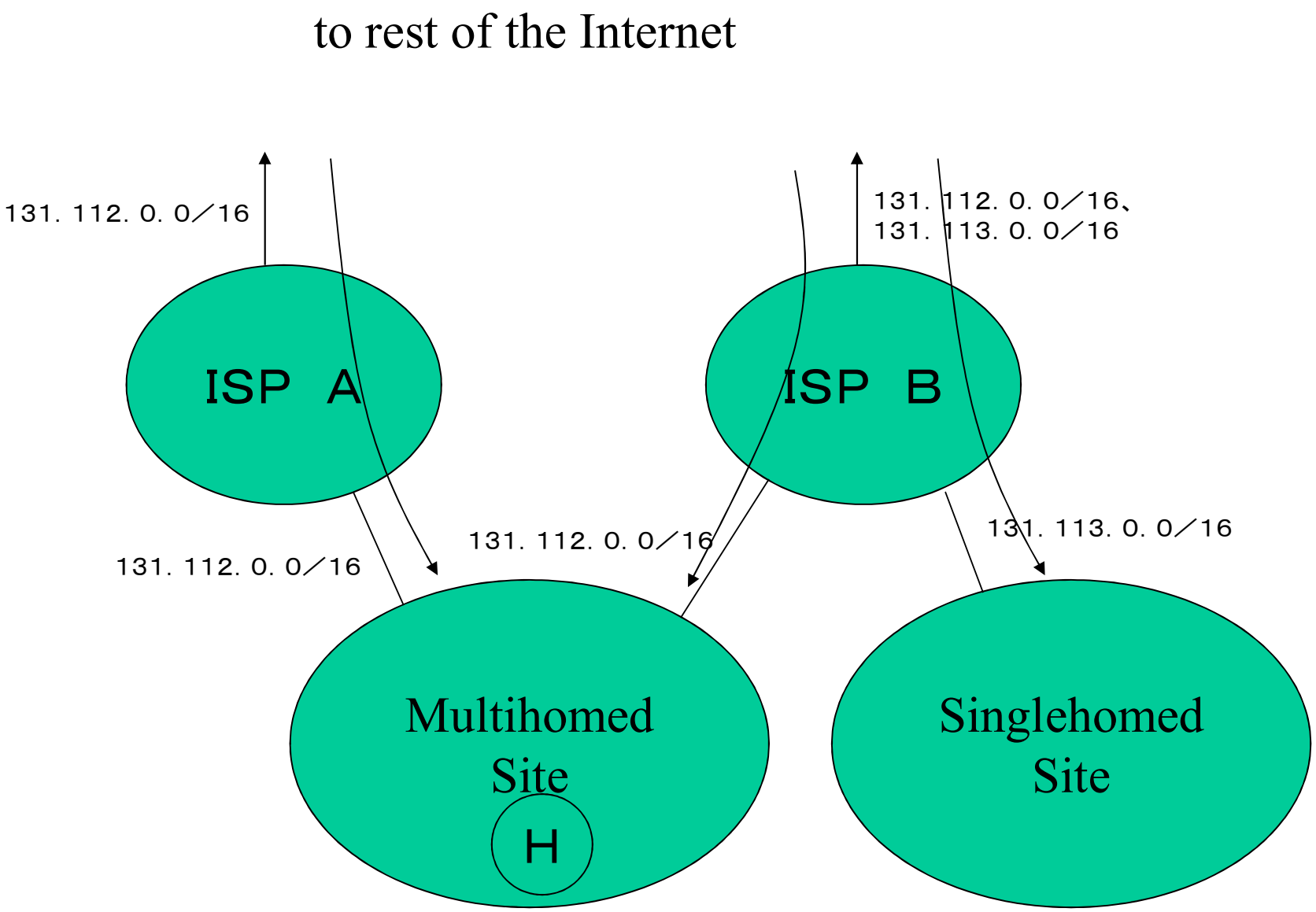
multihoming by routing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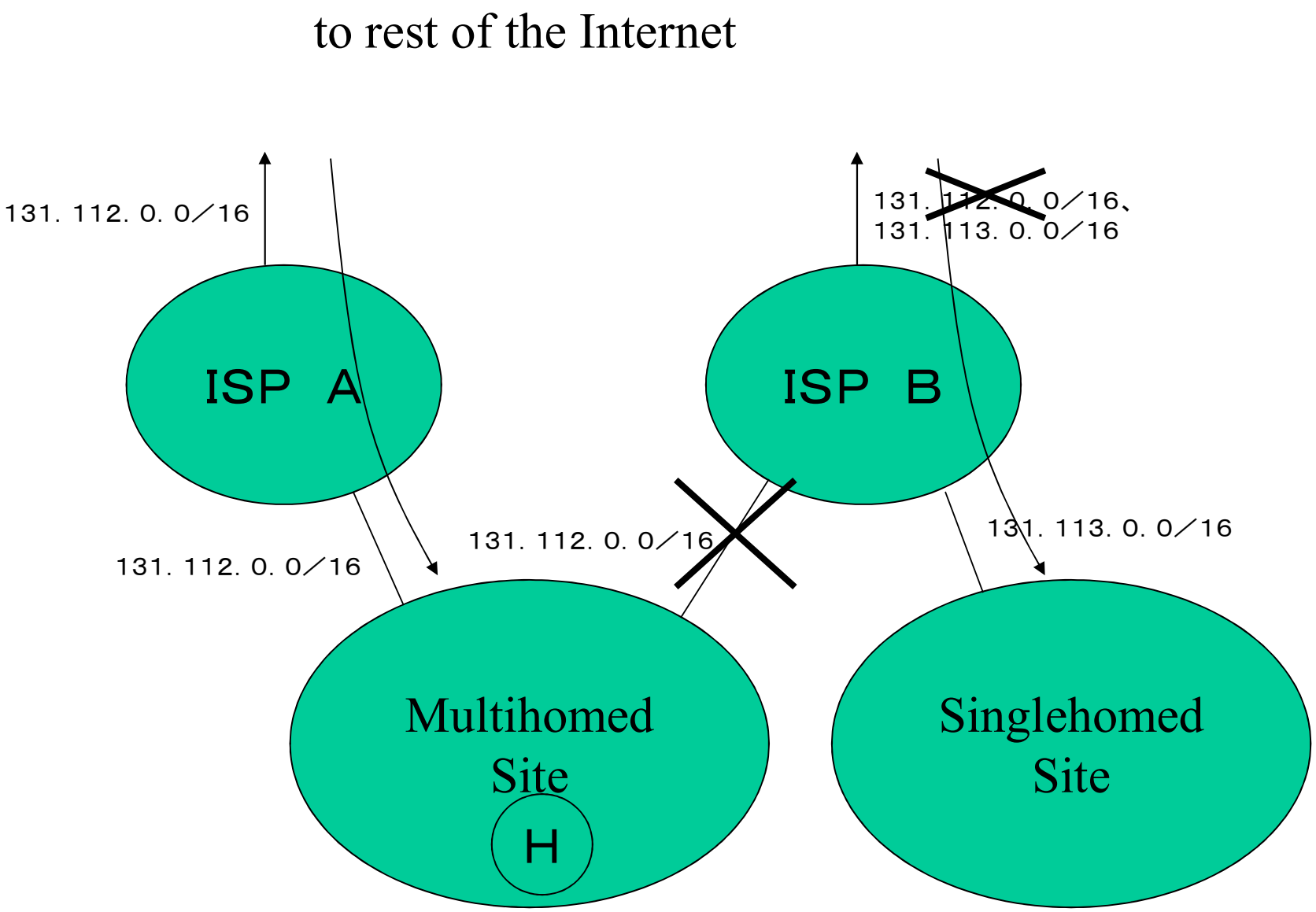
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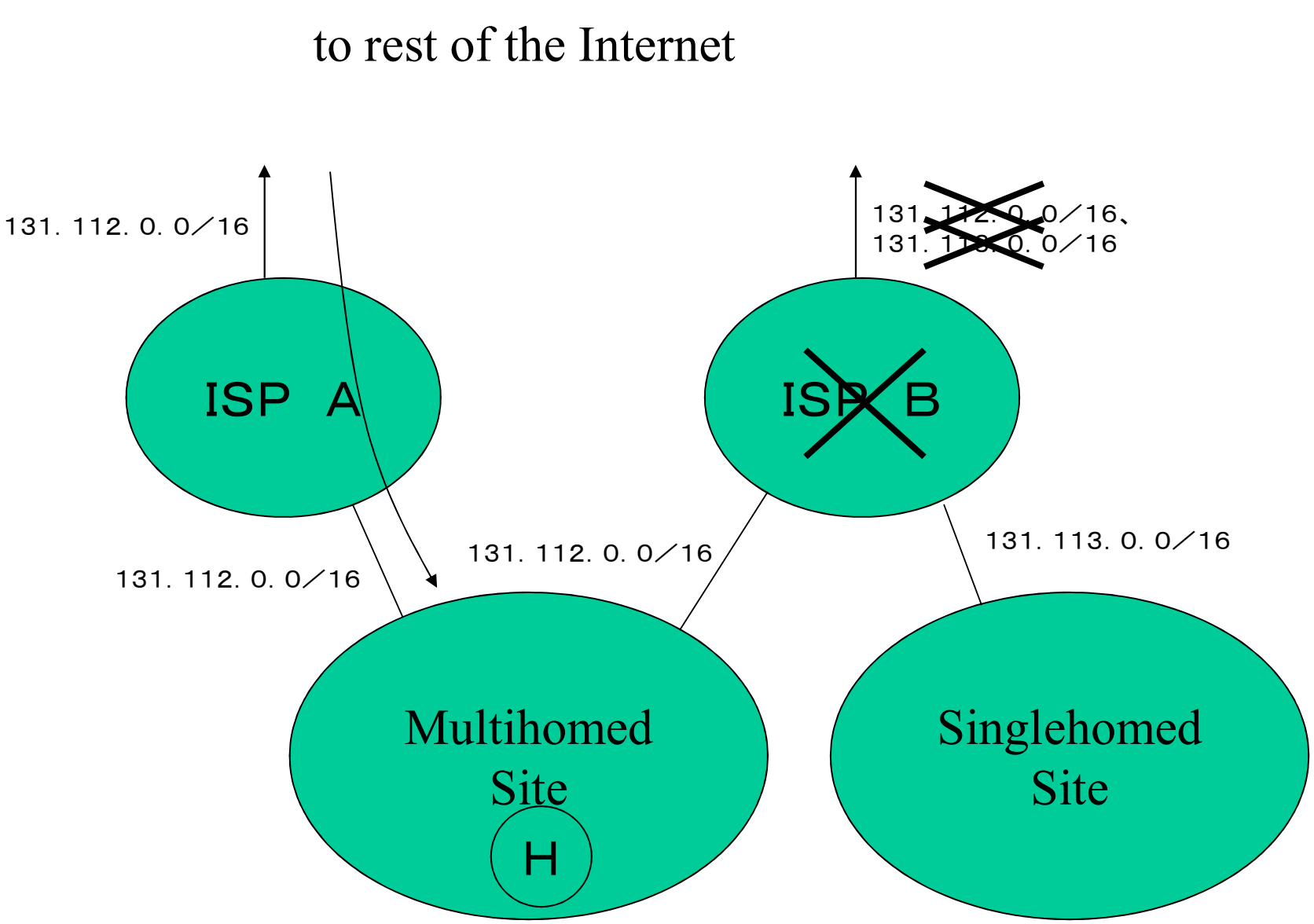
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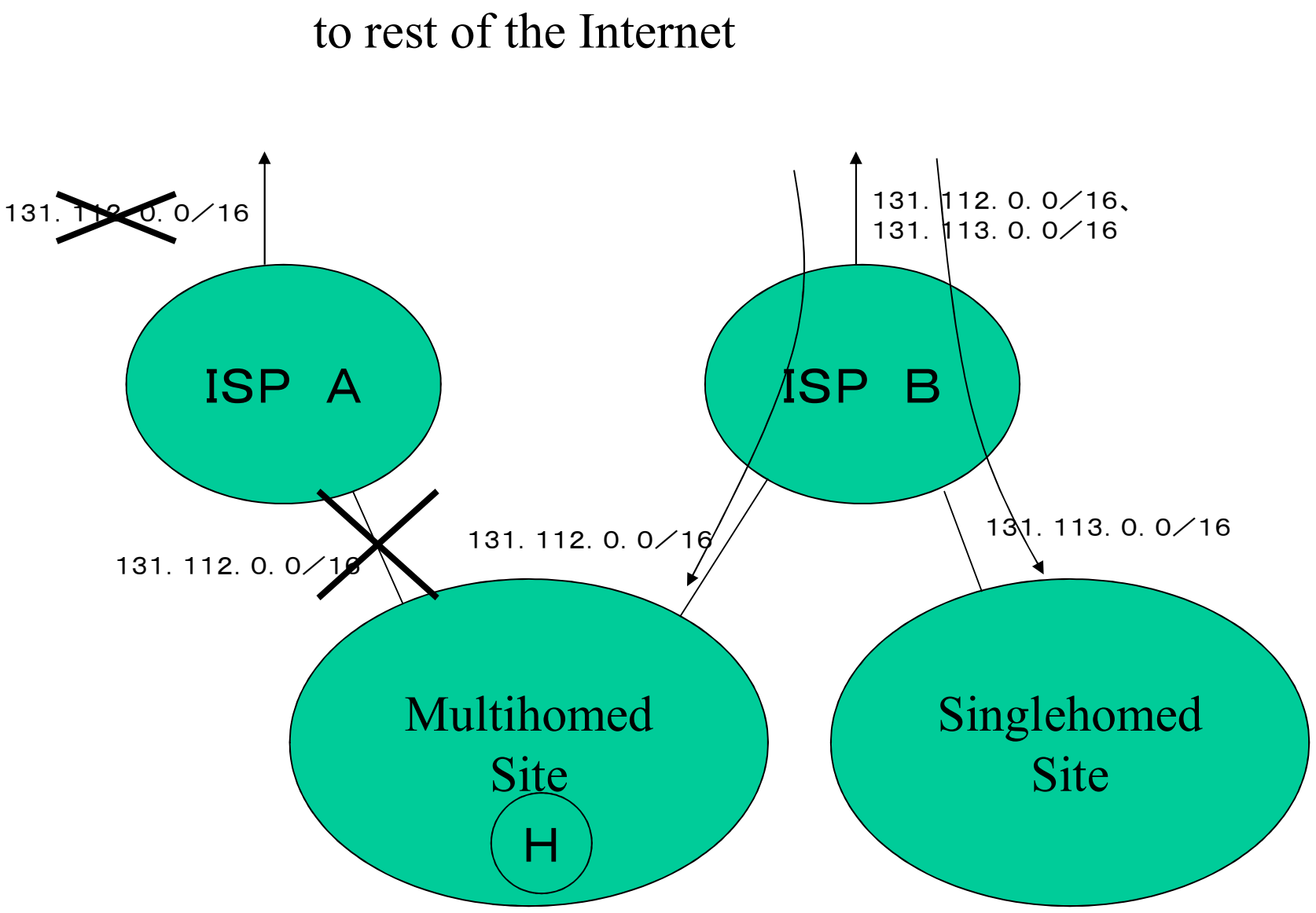
multihoming by routing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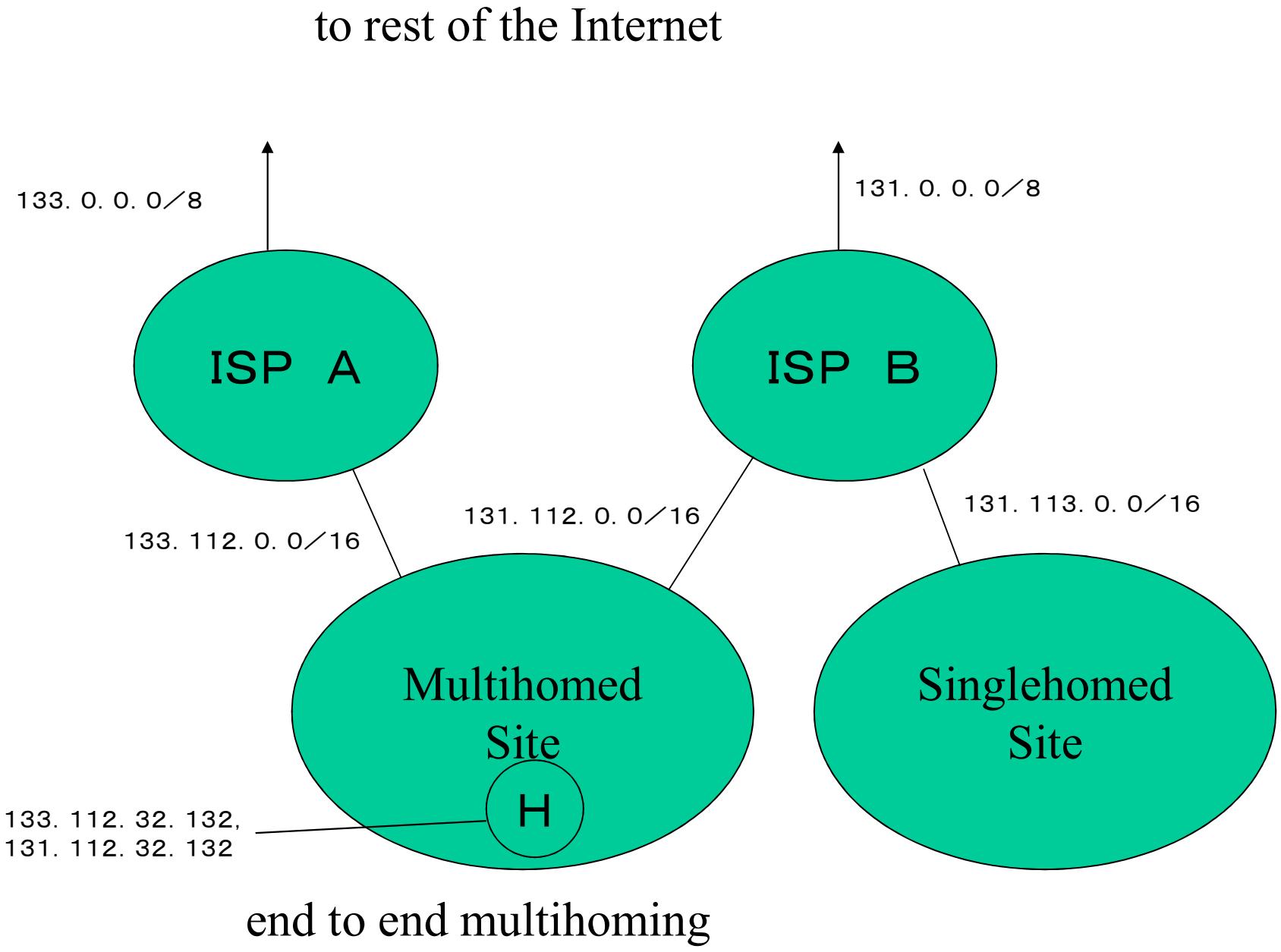
Multihomed
Site
H
Singlehomed
Site
multihoming by routing

End to End Multihoming

- a host has multiple IP addresses
- transport or application layer of peer of the host try to use best address of the host
  - rough unreachability by global routing table
  - if some address works, communication starts
  - if timeout occurs, other addresses are tried
- multihoming by routing is not necessary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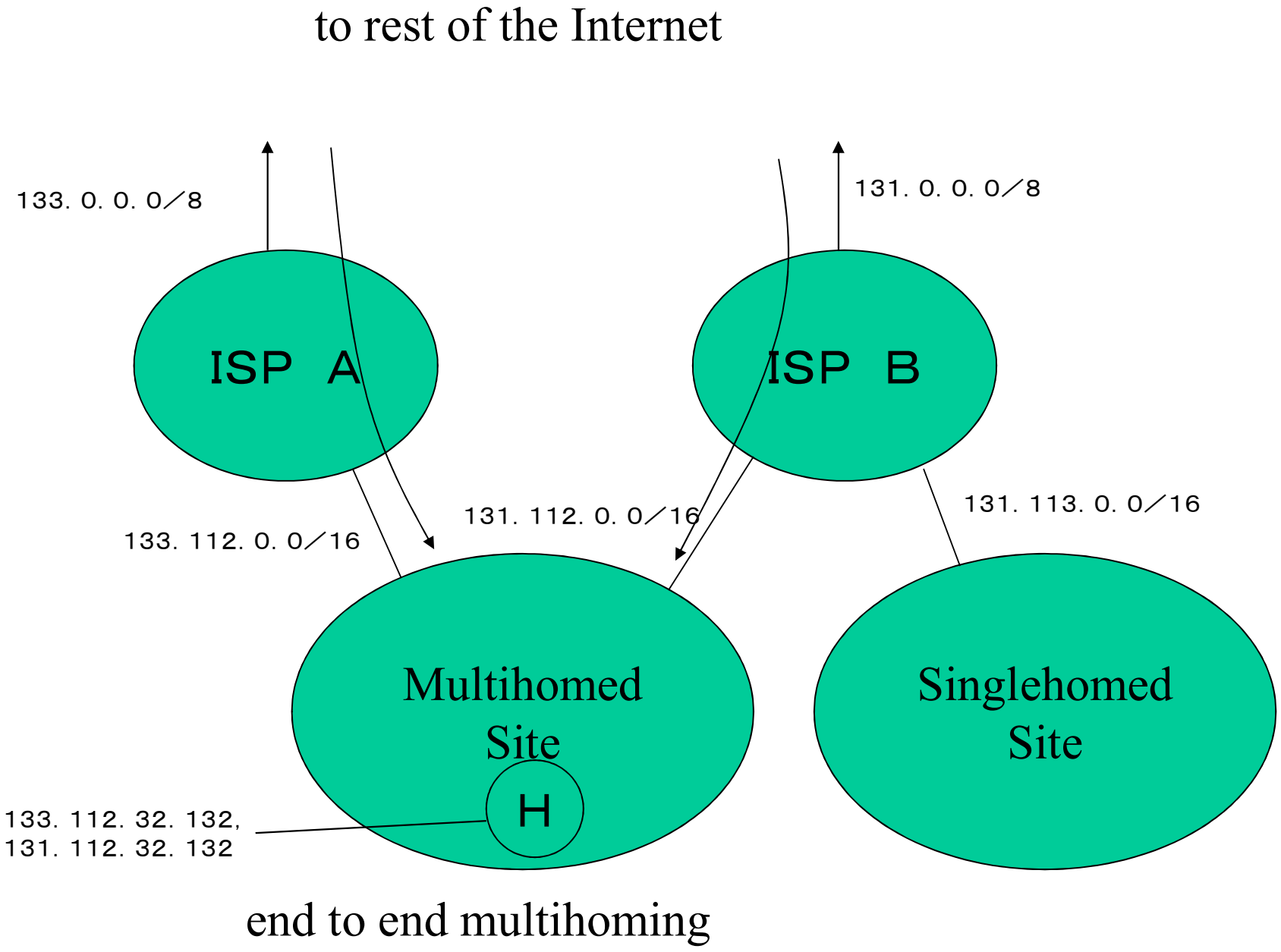
end to end multihoming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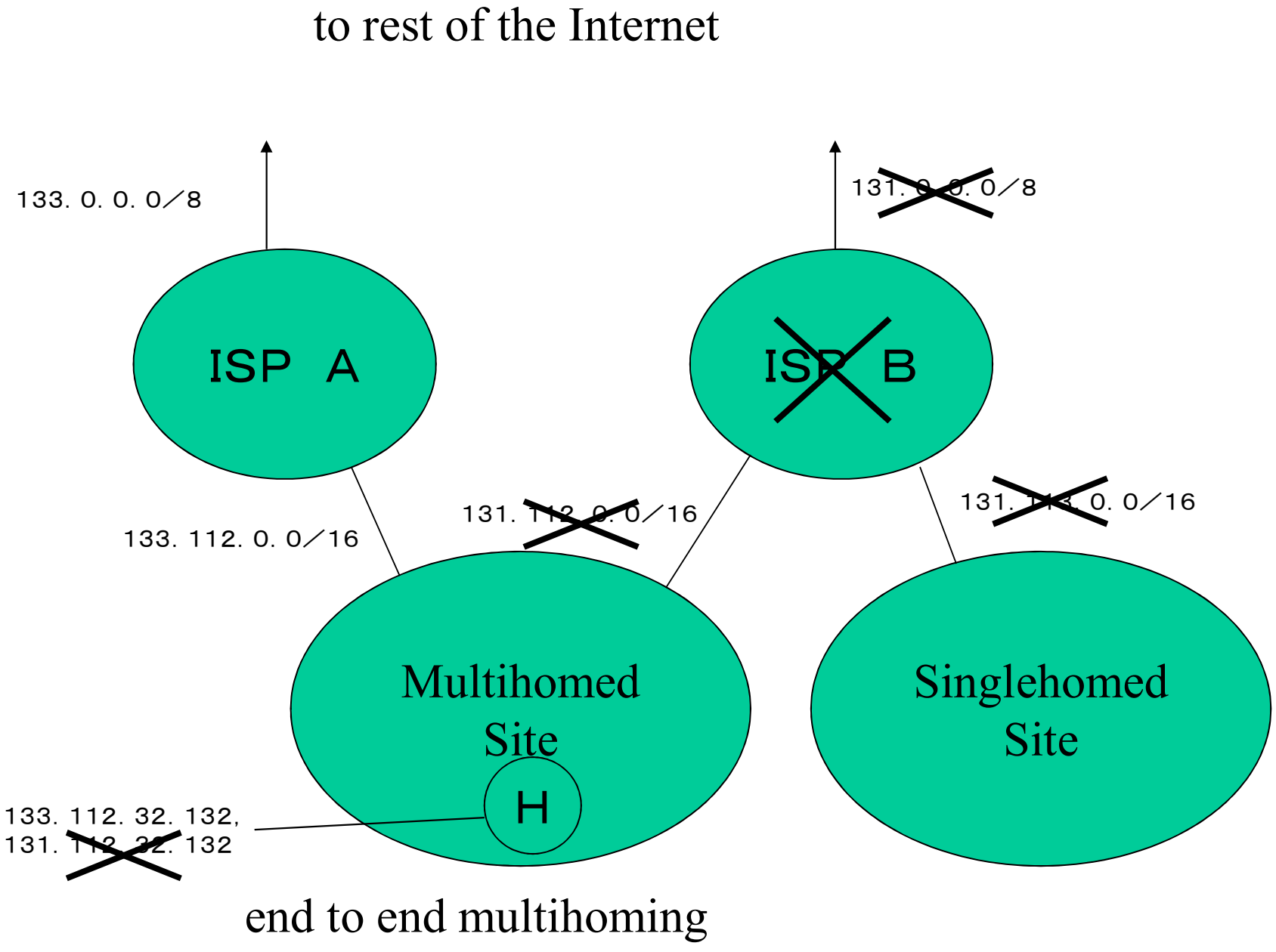
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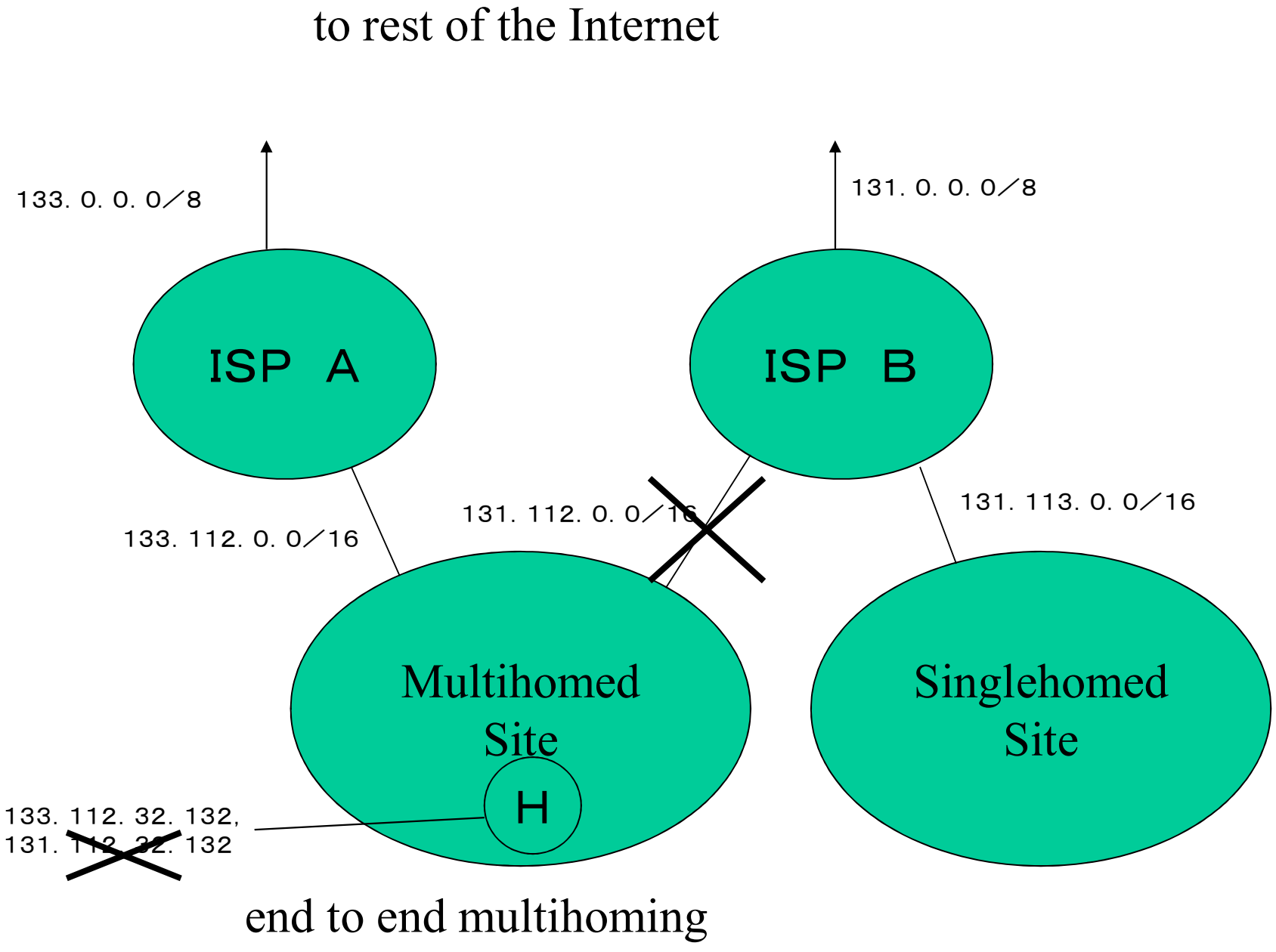
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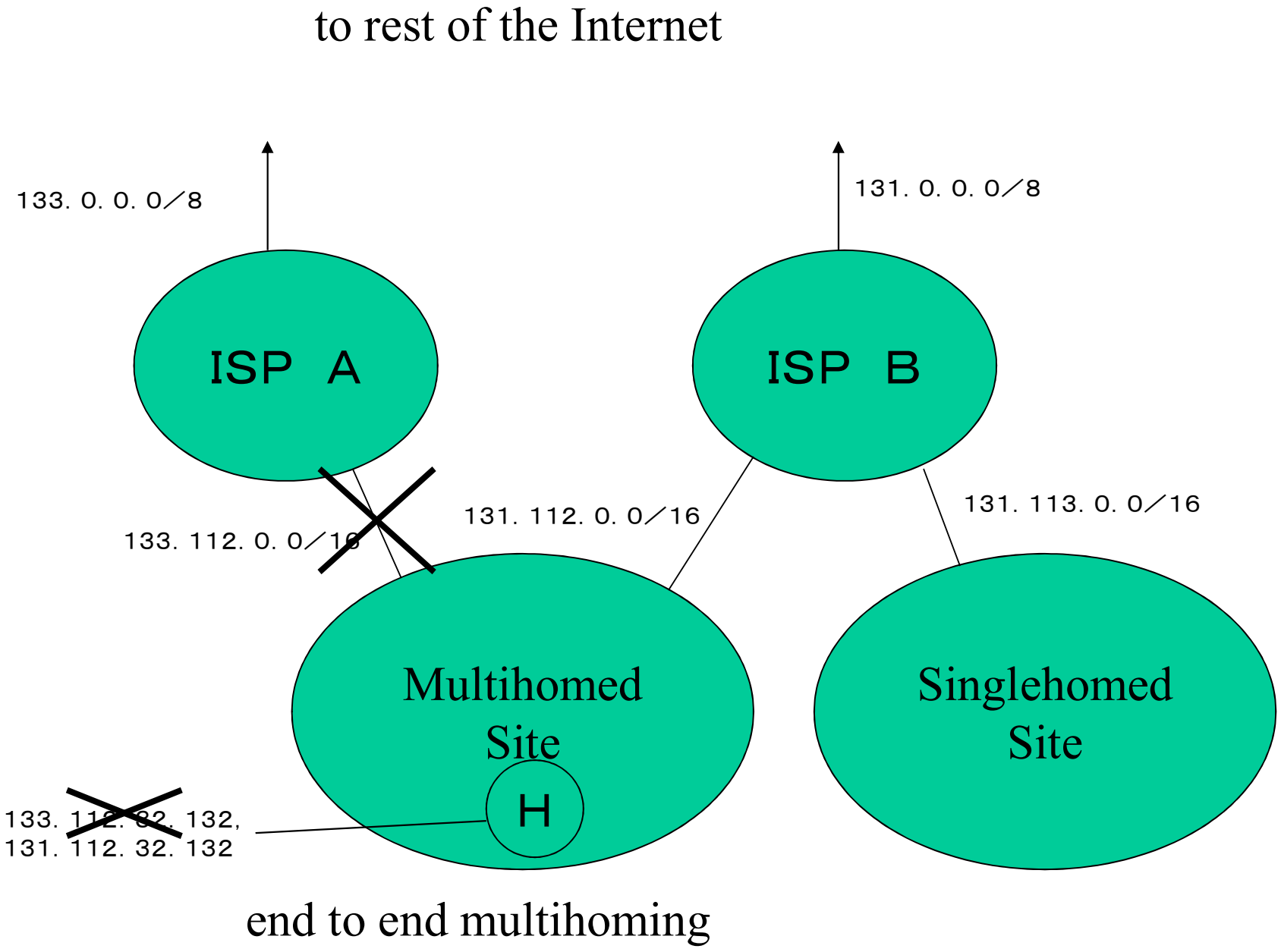
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end to end multihoming

e-mail and E2E Multihoming

- e-mail (SMTP+DNS (rfc974) supports E2E multihoming at application layer
  - if a mail server have multiple addresses
    - all the addresses are tried
  - it is of course as e-mail was the most important application of the Internet
- DNS also support E2E multihoming
  - all the addresses of NSes are tried

Wrap Up

- Internet is as reliable as phone network
- internet does not control BW in network
  - drop packets upon congestion
- dropped packets are retransmitted by TCP
  - speed control is by TCP at ends
- multihoming should also be performed by ends