Common network/protocol functions

Goals:

- Identify, study common architectural components, protocol mechanisms
- *Synthesis:* big picture
- *Depth:* important topics not covered in introductory courses

Overview:

- Signaling: TCP, Telephone network, Internet
  - Protocols
- State handling
- Randomization
- Indirection
- Service location
- Network virtualization

Common but already covered …

- Flow control
- Congestion control
- Routing
- Timers
- Naming and addressing
- others?

Signaling

Signaling: exchange of messages among network entities to enable (provide service) to connection/call

- Before, during, after connection/call
  - Call setup and teardown
  - Call maintenance
  - Measurement, billing
- Between
  - End-user <-> network
  - End-user <-> end-user
  - Network element <-> network element

Examples

- Internet
  - TCP handshake
  - RSVP (Resource Reservation Protocol)
  - SIP
- Telephone network
  - SS7 (Signaling System no. 7)

Signaling in the Internet

connectionless (stateless) forwarding by IP routers + best effort service = no network signaling protocols in initial IP design

- New requirement: Transport protocol needs state and variable initialization
- TCP: Transport Control Protocol [RFCs 793, 1122, 1323, 2018, 2581]

TCP Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- Initialize TCP variables:
  - Seq. #s
  - Buffers, flow control info (e.g. **`RcvWindow`**)
  - MSS and other options
- *Client:* connection initiator, *server:* contacted by client

- Three-way handshake
  - Simultaneous open
- TCP Half-Close (four-way handshake)
- Connection aborts via RSTs

TCP Connection Management (2)

Three way handshake:

Step 1: Client end system sends TCP SYN control segment to server

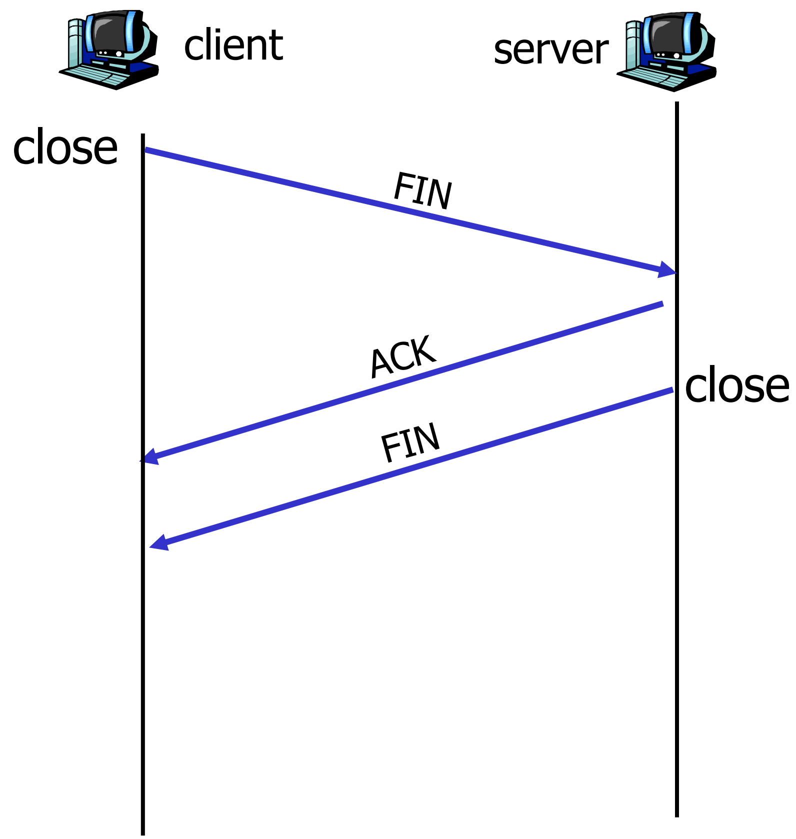
- Specifies initial seq #
- Specifies initial window #

Step 2: Server end system receives SYN, replies with SYNACK control segment

- ACKs received SYN
- Allocates buffers
- Specifies server → receiver initial seq. #
- Specifies initial window #

Step 3: Client system receives SYNACK, sends ACK

TCP Connection Management (3)



Closing a connection:

Client closes socket:

```
clientSocket.close();
```

Step 1: Client end system sends TCP FIN control segment to server

Step 2: Server receives FIN, replies with ACK. Closes connection, sends FIN.

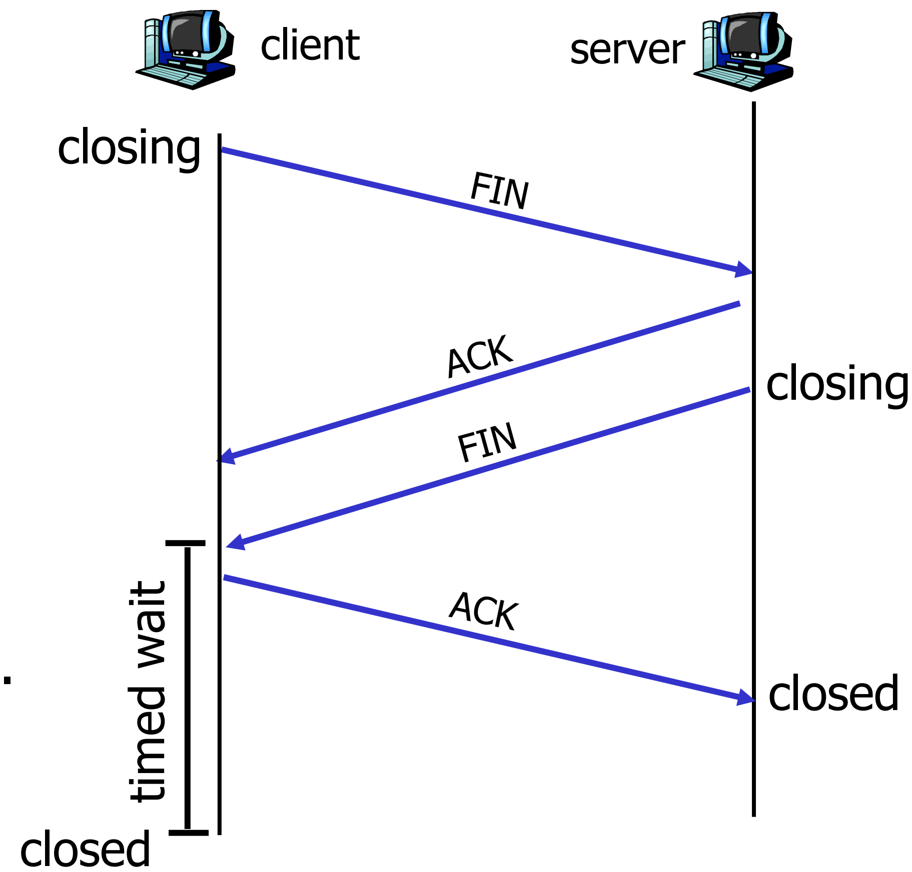
TCP Connection Management (4)

Step 3: Client receives FIN, replies with ACK.

- Enters "timed wait" – will respond with ACK to received FINs

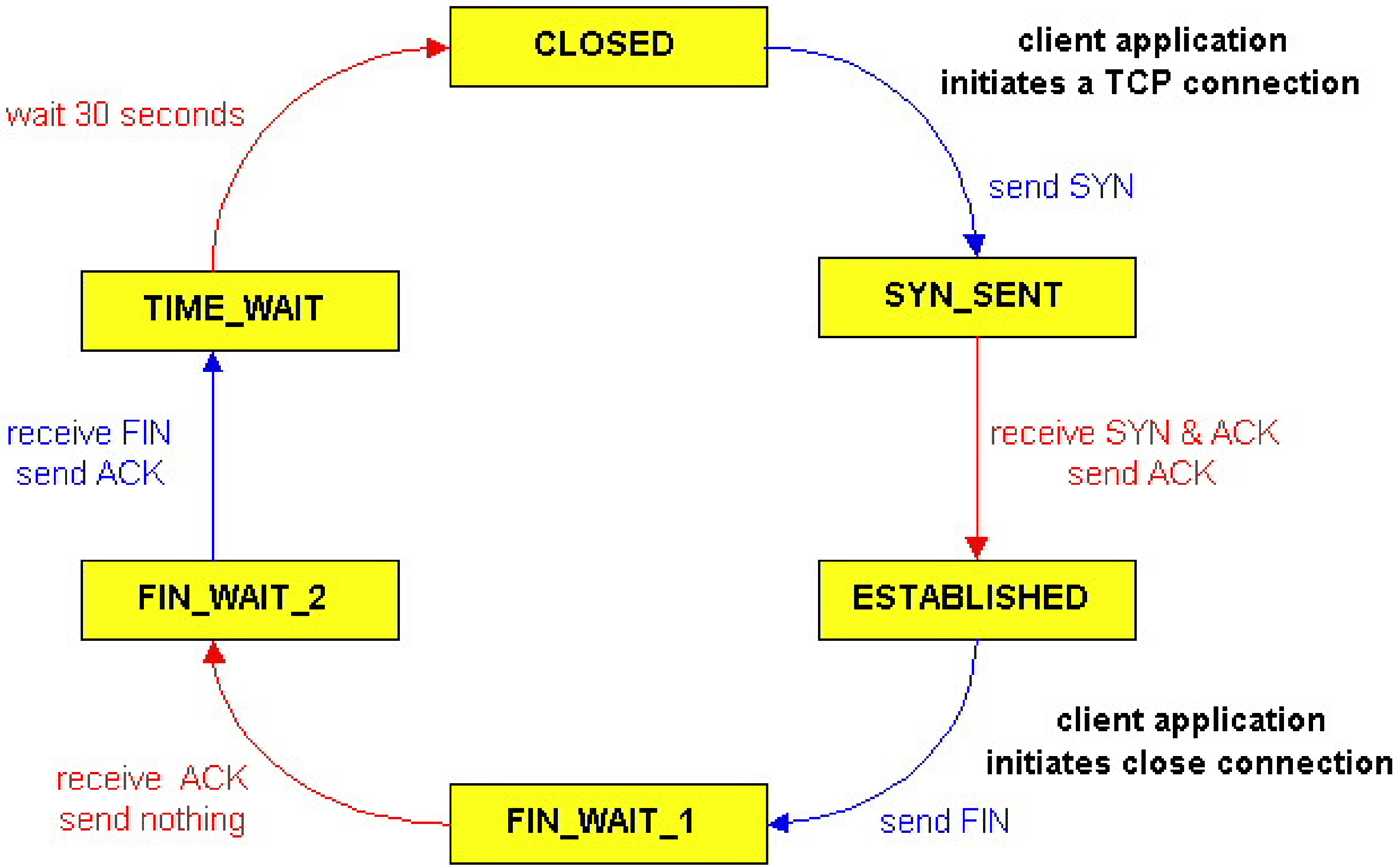
Step 4: Server, receives ACK. Connection closed.

Note: With small modification, can handly simultaneous FINs.



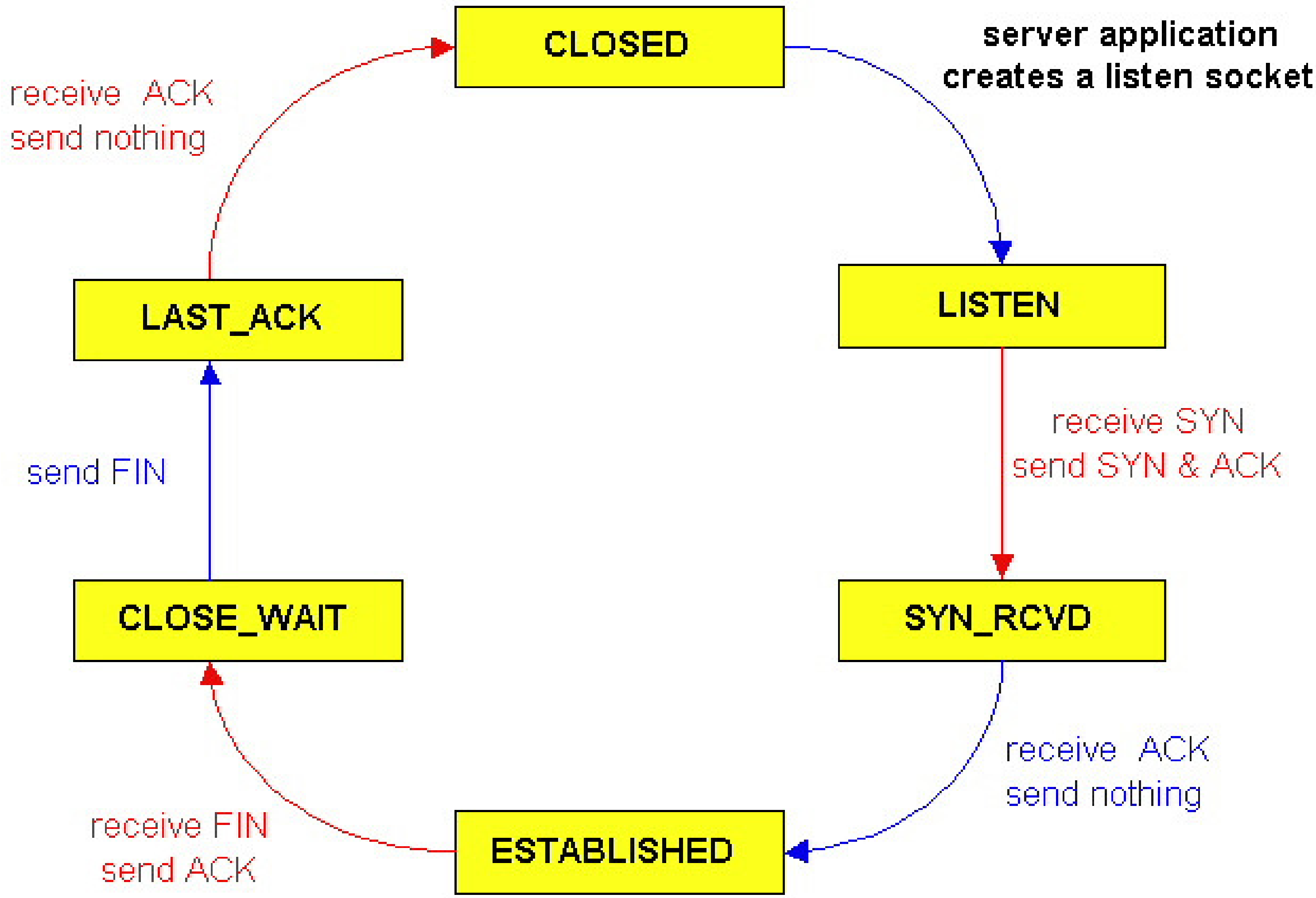
TCP Connection Management (5)

TCP client lifecycle

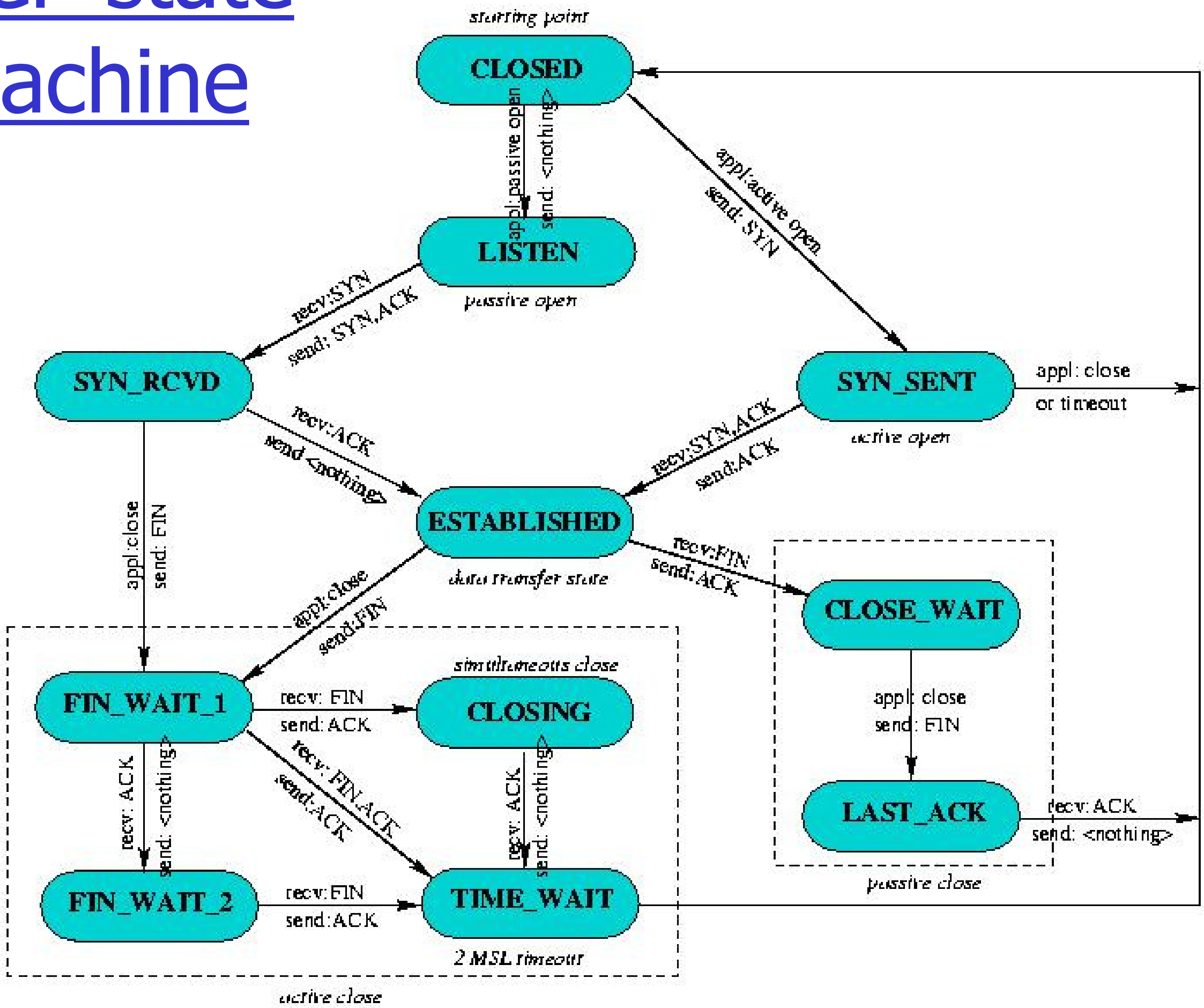


TCP Connection Management (6)

TCP server lifecycle

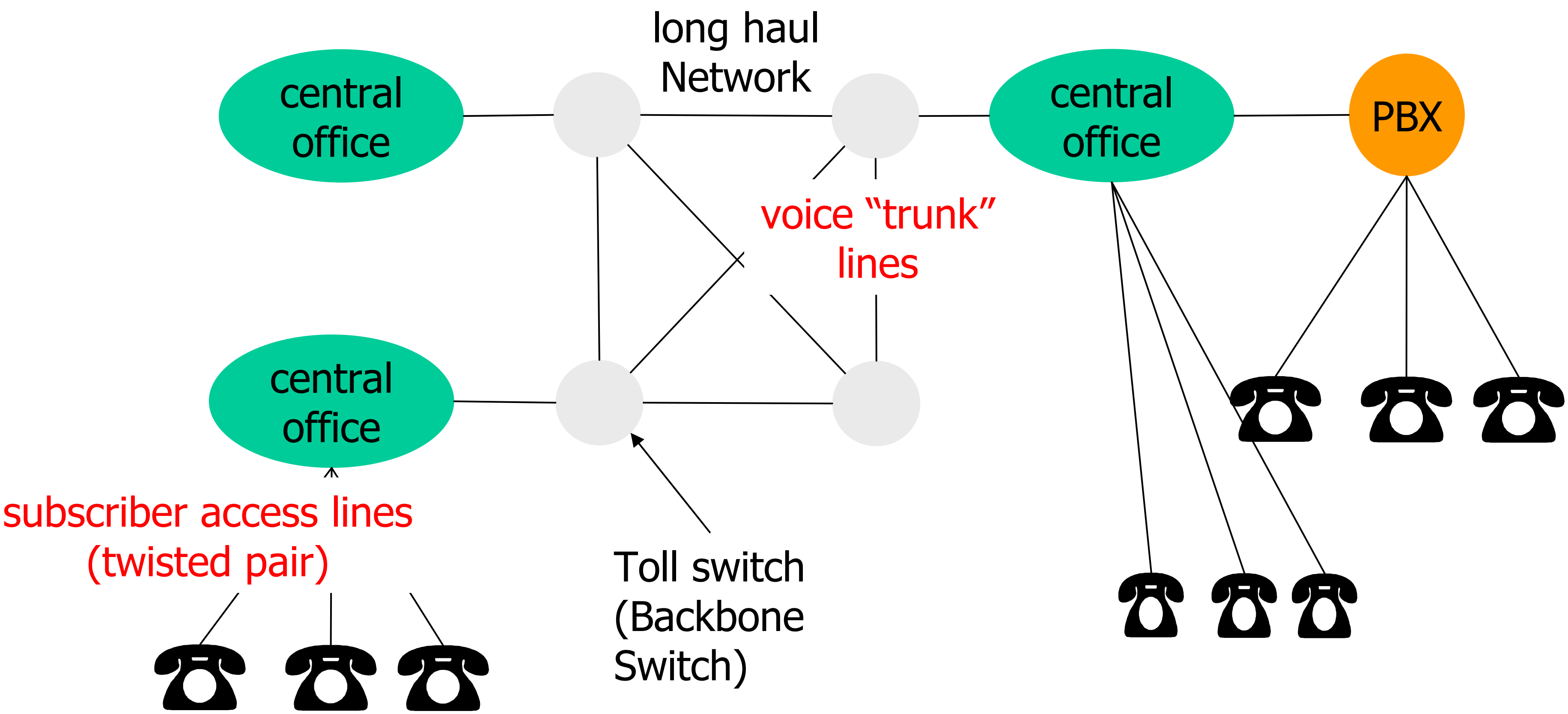


TCP state machine

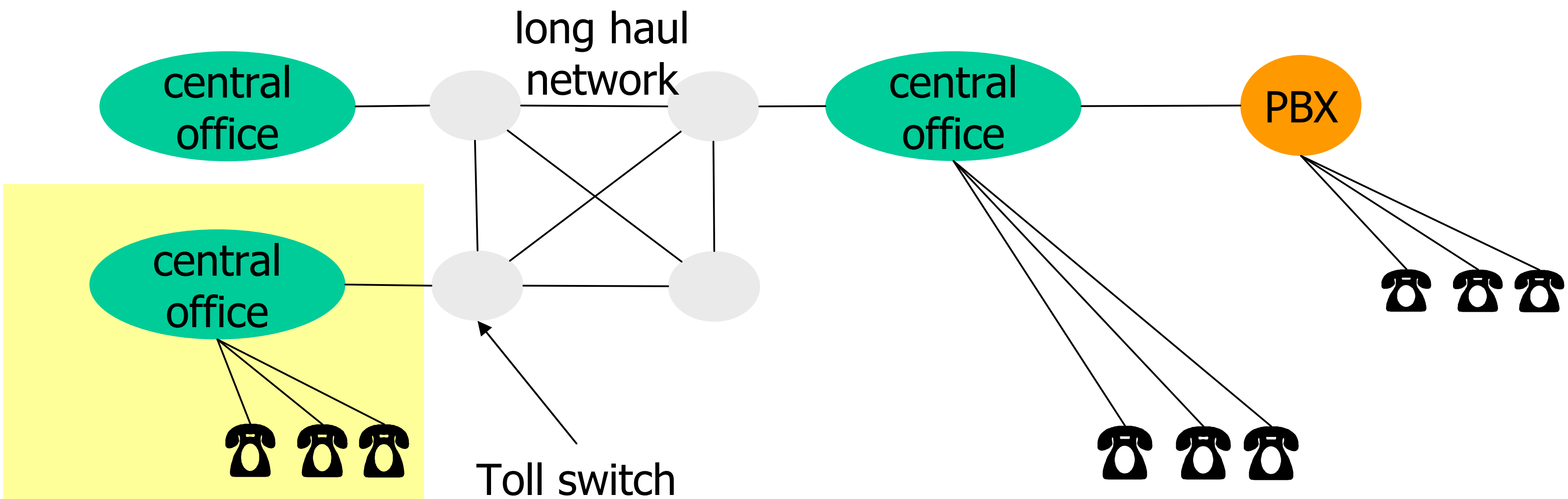


Telephone network

- Created 1876
- Currently a global Infrastructure

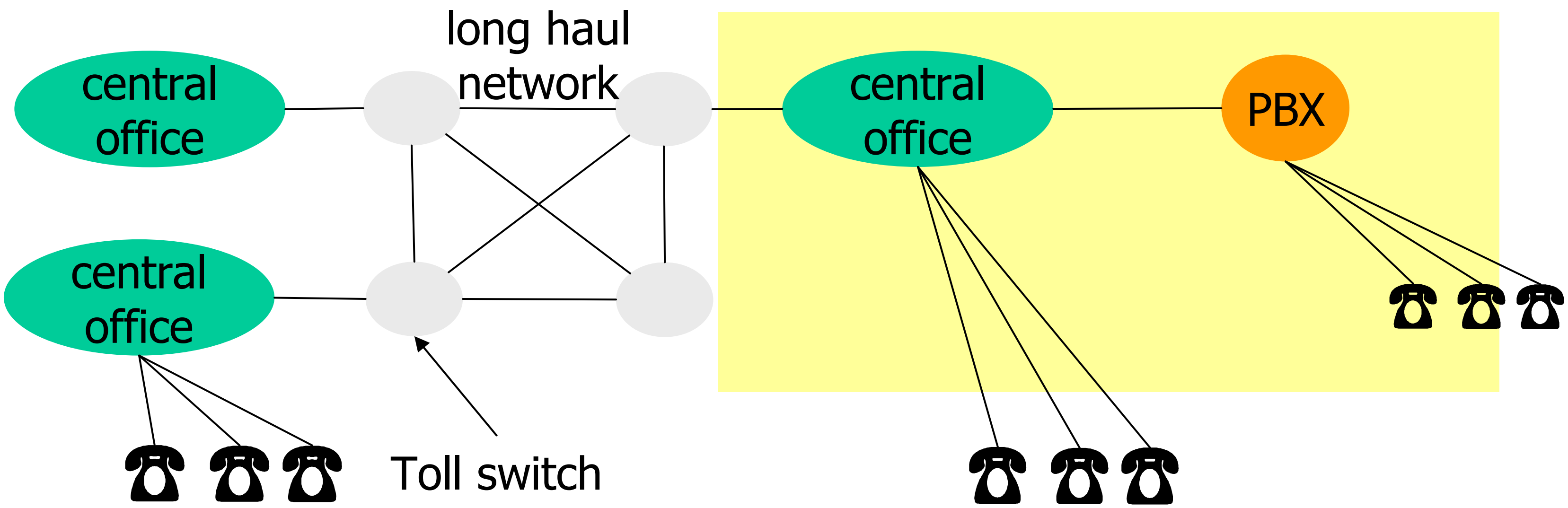


Central office and local loop



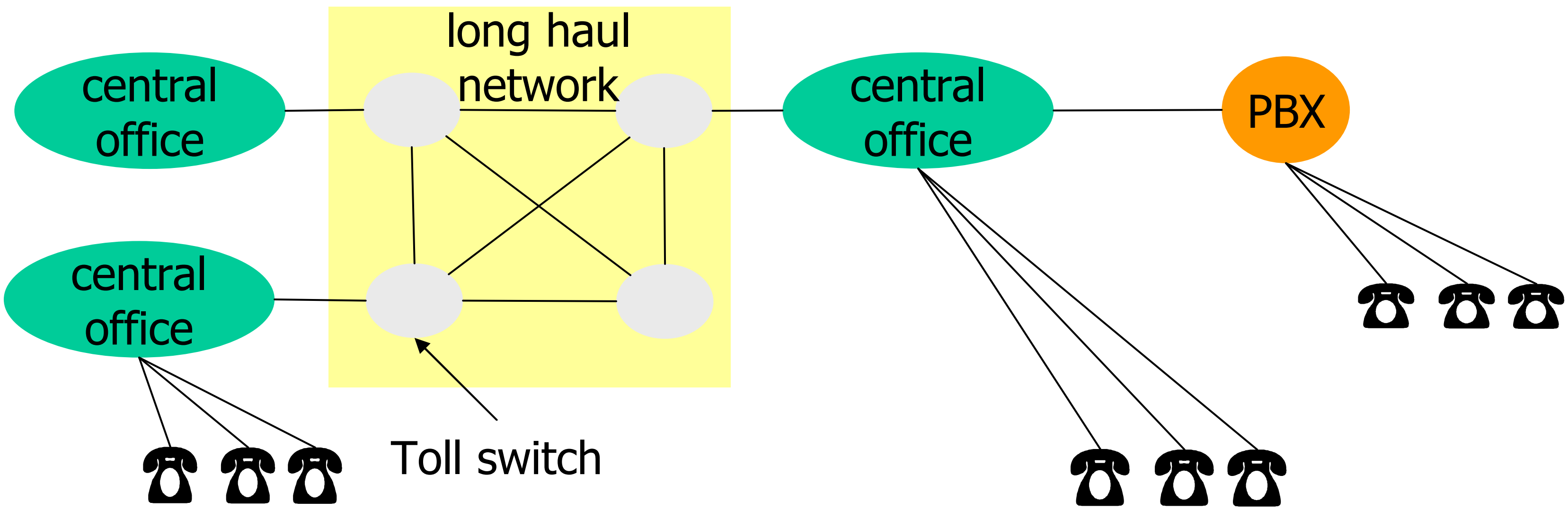
- Each phone user (**subscriber**) has direct connection to switch in central office (**local loop)**
- Local loop has length 1 - 10 km
- Switches in central office called **(local) exchanges**
- Company providing local telephone service called **local exchange carrier** or **LEC** (e.g., Bell Atlantic)

Private Branch Exchange (PBX)



- **PBX (Private Branch Exchange)** telephone system within enterprise that switches calls on local lines; allows users to share fixed number of external lines to central office
- Saves cost of line per user to central office

Long-haul network

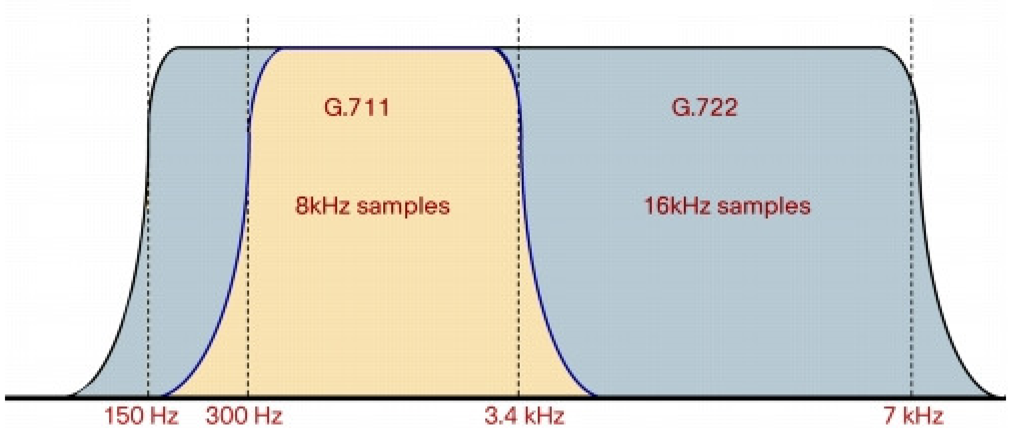


- Toll switches provide long-distance connectivity over long distance trunks
- ~500 toll switches in US
- Toll switch runs 100,000^{+} phone calls

How is voice transmitted ?

Two ways:

- Analog voice transmission: voice channel allocated bandwidth of 3.5 kHz
- Digital voice transmission: analog voice stream converted to digital stream
  - Standard scheme: 8000 8 bit samples == 64Kbps
  - Wide band: e.g., 16.000 8 bit samples



The digital phone network

Until 1960s:

- Analog telephone network
- Frequency-division multiplexing

Today:

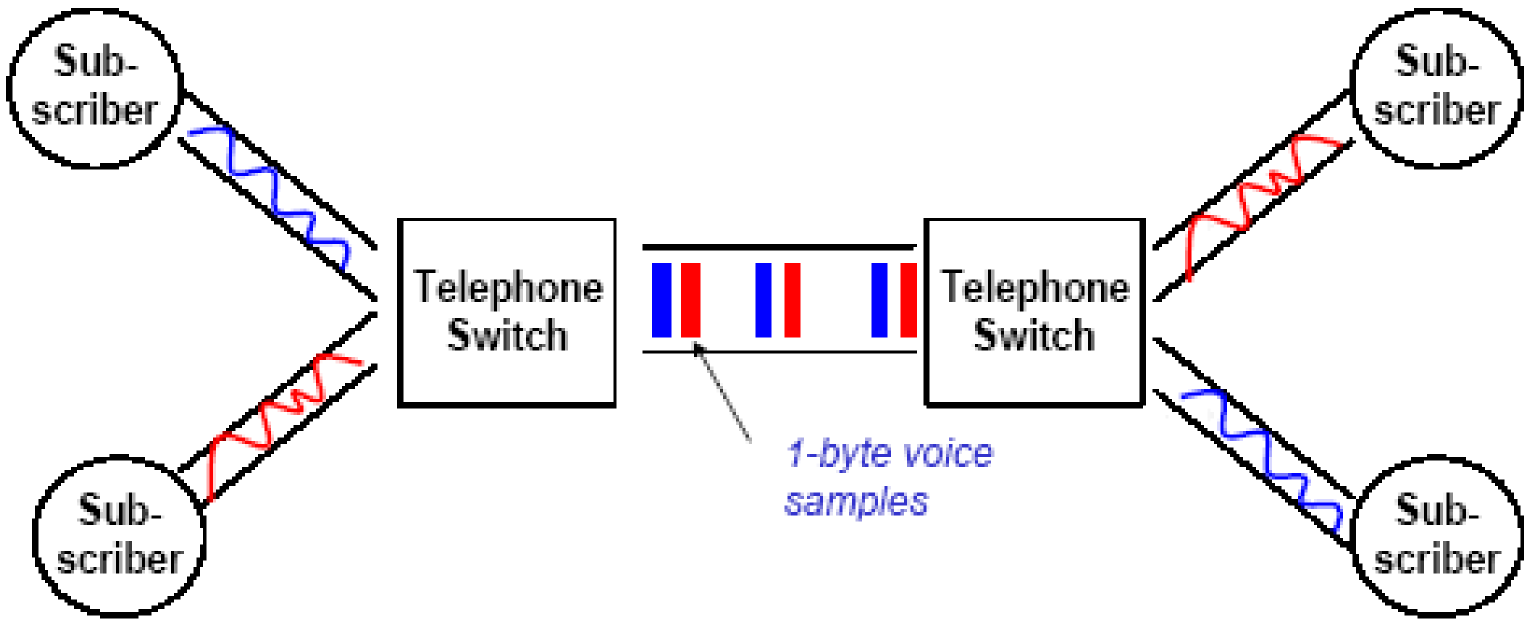
- Local loop analog
- ISDN (Integrated services Digital Network) all digital circuit switching technology. Available since the early-1990s (in Europe) or mid-1990s (US). No wide deployment in US
- Rest of network digital (based on TDM)

When do we get all digital network?

- All ISDN (No wide deployment in US)

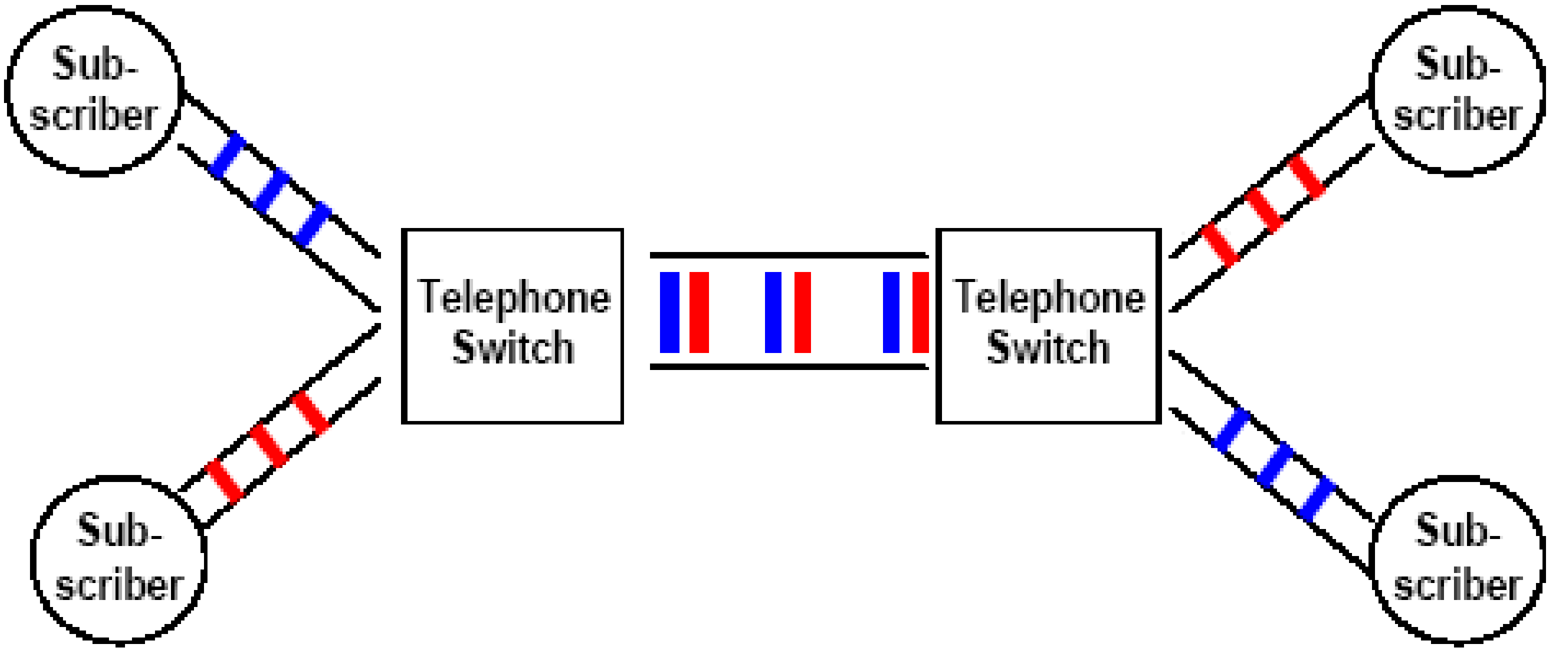
Another all digital – but **not** circuit-switched – telephony solution is IP telephony

Analog Loop / Digital Network



- First telephone switch digitizes voice call (8000 8-bit samples per second)
  - Switching method is TDM.
- Switch multiplexes calls, interleaving samples in time. call receives one 8-bit slot every 125 µs

All Digital Network



- Telephone at subscriber digitizes voice, sends one 8-bit sample every 125 μs

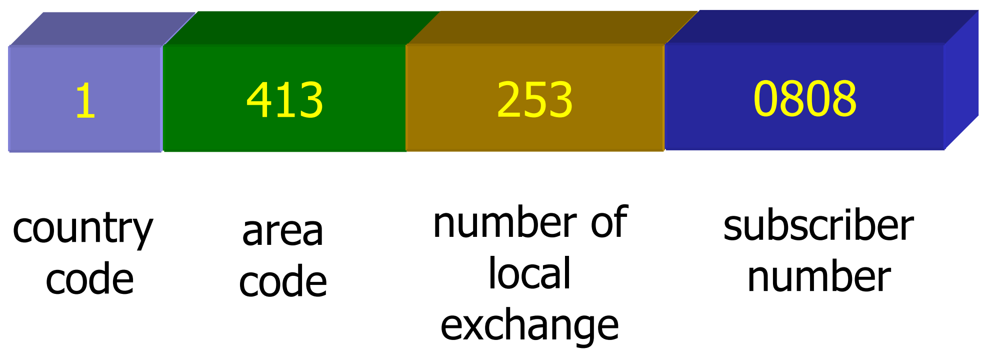
Digital Multiplexing

- Digital Signaling (DS) transmission hierarchy used in US for multiplexing digital voice channels

	Number of voice circuits	Bandwidth
DS0	1	64 kbps
DS1	24	1.544 Mbps
DS2	96	6.312 Mbps
DS3	672	44.736

Addressing and routing

- Each subscriber has address (telephone number)
  - Hierarchical addresses
  - Example: Antonio's Pizza in downtown Amherst



- Telephone address used for setting up route from caller to callee

Telephone network: Services

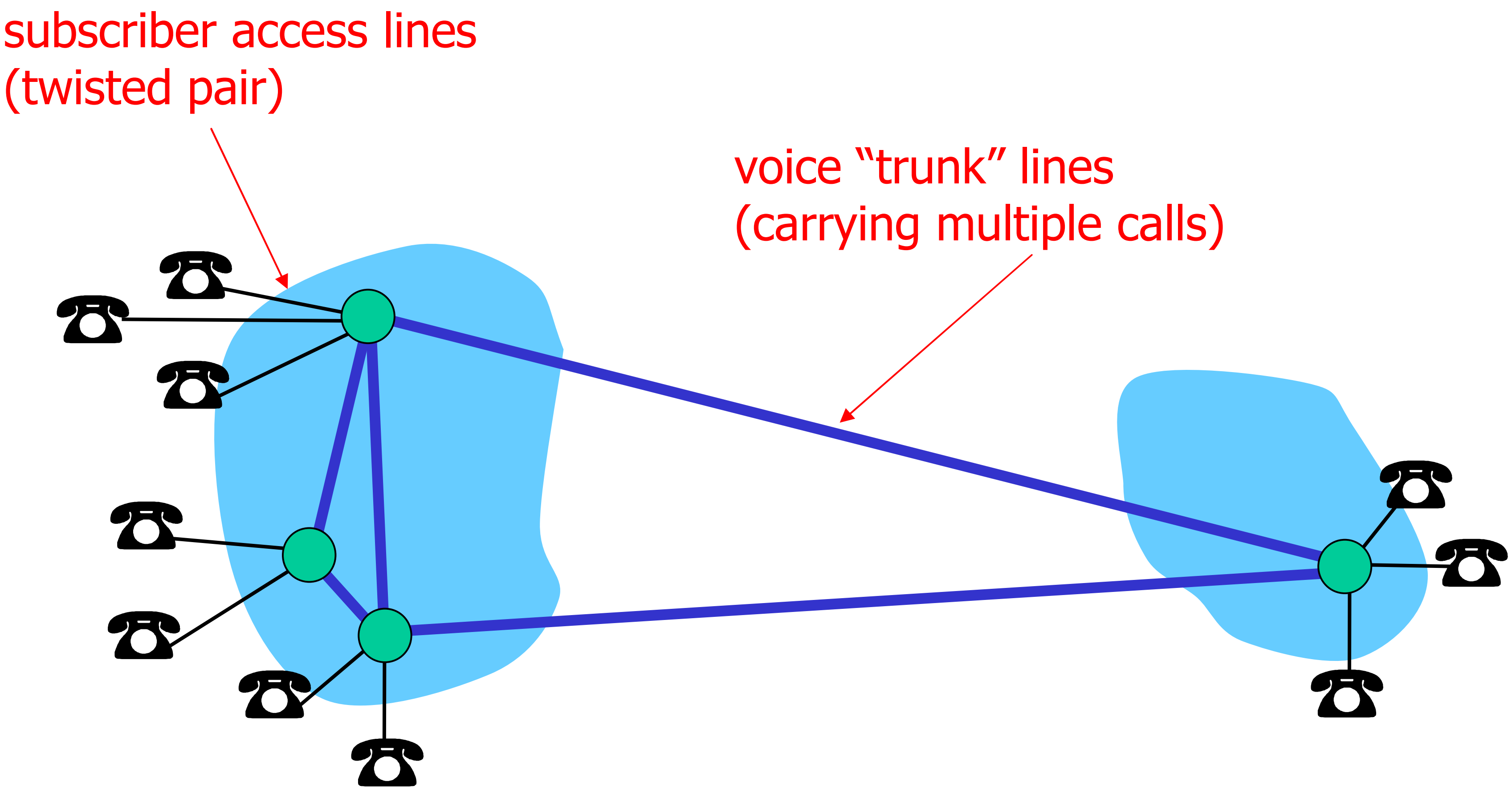
- Point-to-point POTS calls
- Special telephone numbers
  - 800 (888) number service: Free call to customer
  - 0180 number service: Fixed fee call for customer
  - 900 number service: Bill caller
  - Numbers for life
- Caller ID
- Calling card/third part charging
- Call routing (to end user):
  - Prespecified by time-of-day
- "Follow me" service
- Incoming/outgoing call restrictions
- Support for cellular roaming:
  - "Home" number routed to current cell location

Telephone network: AIN

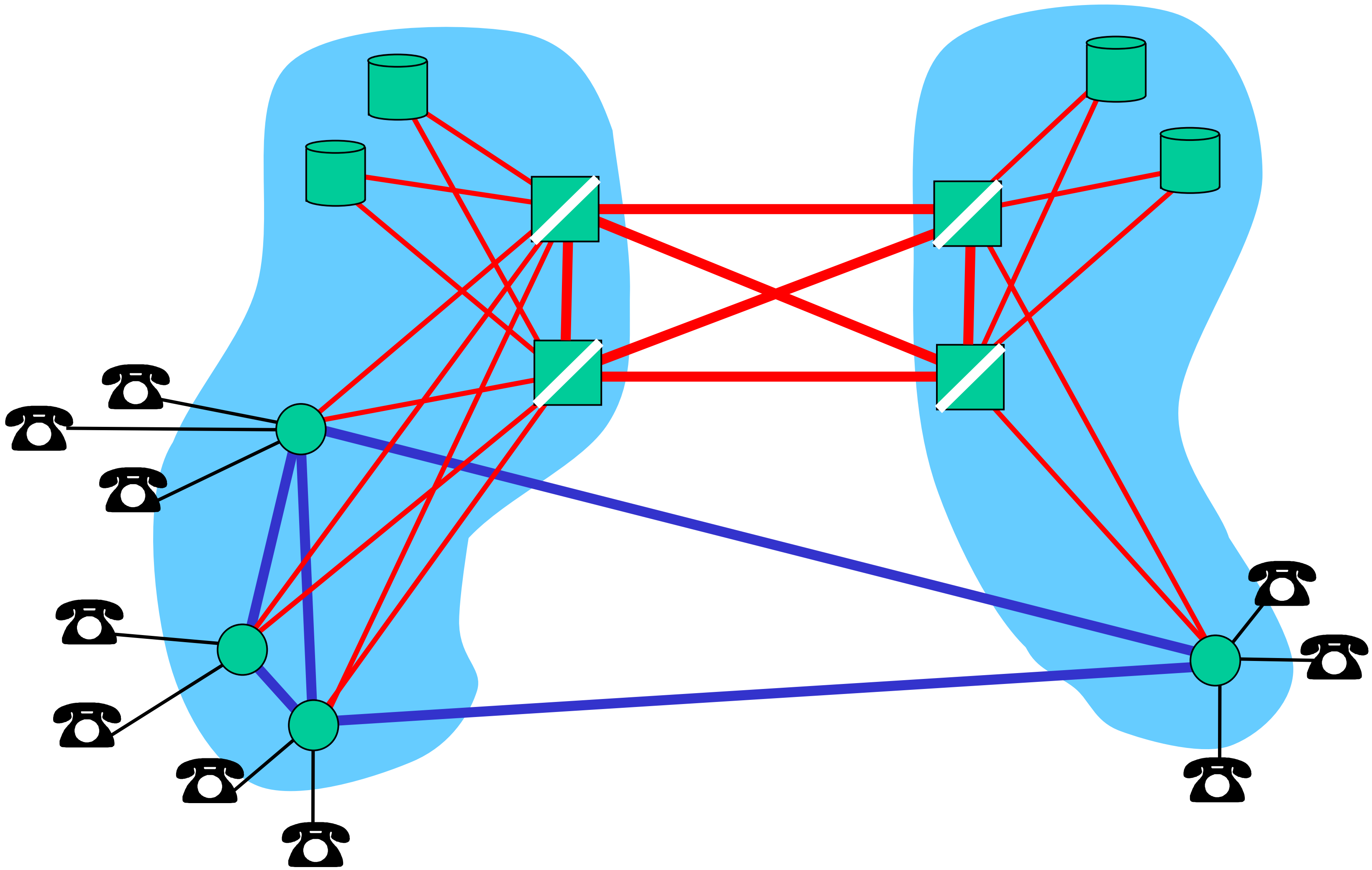
AIN: Advanced Intelligent (phone) Network: Migration from service-in-the-switch to service logic external to (on top of) switching systems

- Looks like Internet philosophy:
  - E.g., DNS is at application layer; RIP,OSPF, BGP above IP
- AIN advantages:
  - Introduce new services rapidly
  - Open interfaces
    - Vendor customization
    - Vendor independence of services

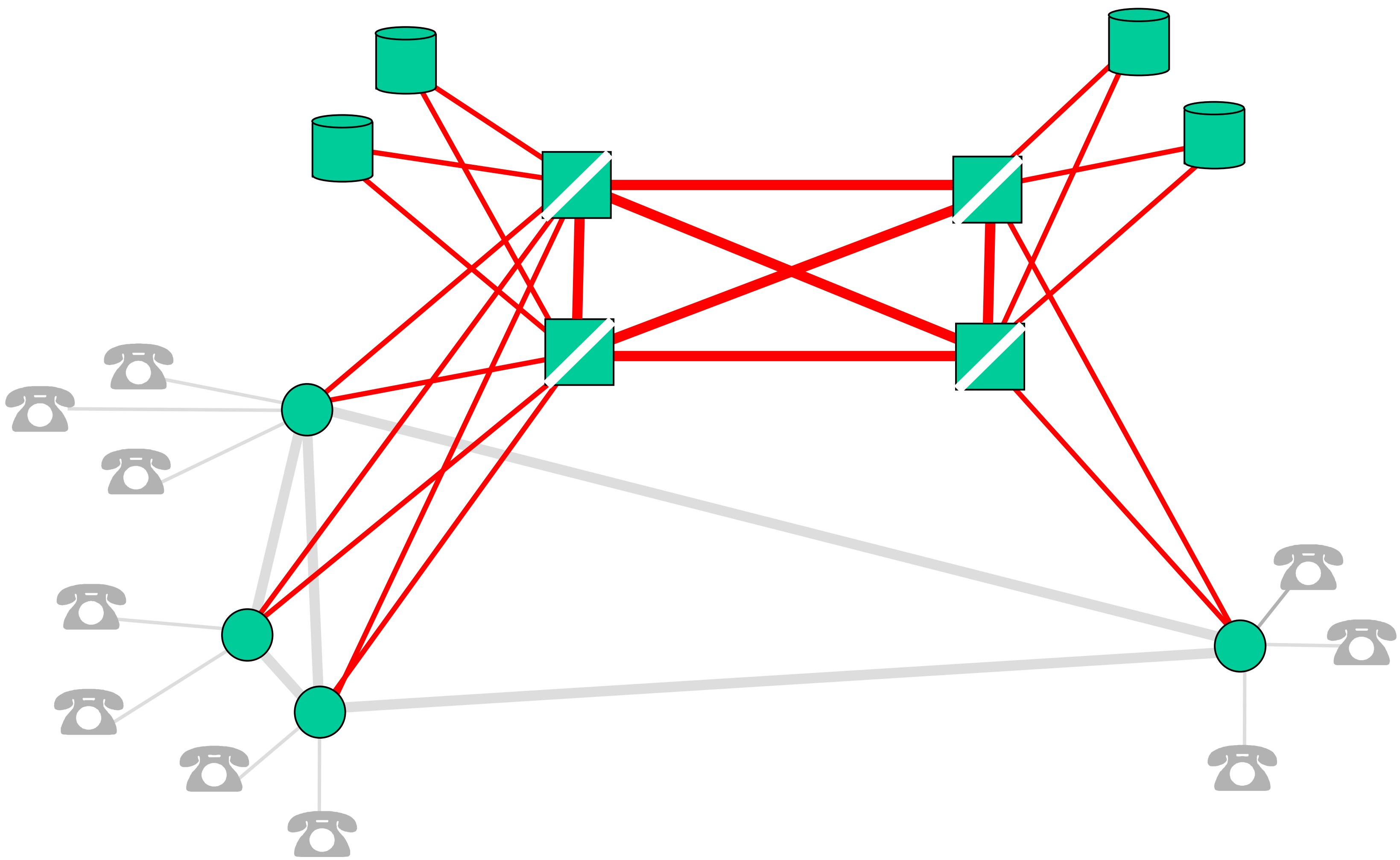
Telephone network: Circuit-switched voice trunks (data plane)



Telephone: Data and control plane



SS7: Telephone signaling network



SS7: Telephone network signaling

- *Out-of-band signaling:* telephony signaling carried over *separate* network from telephone calls (data)
  - Allows for signaling between any switches (not just directly-connected)
  - Allows for signaling during call (not just before/after)
  - Allows for higher-than-voice-data-rate signaling
  - Security: In-band tone signaling helps phone phreaks; out of band signaling more secure
- Signaling System 7 (SS7) network: *Packet-switched*
  - calls themselves circuit-switched
- Lots of redundancy (for reliability) in signaling network links, elements

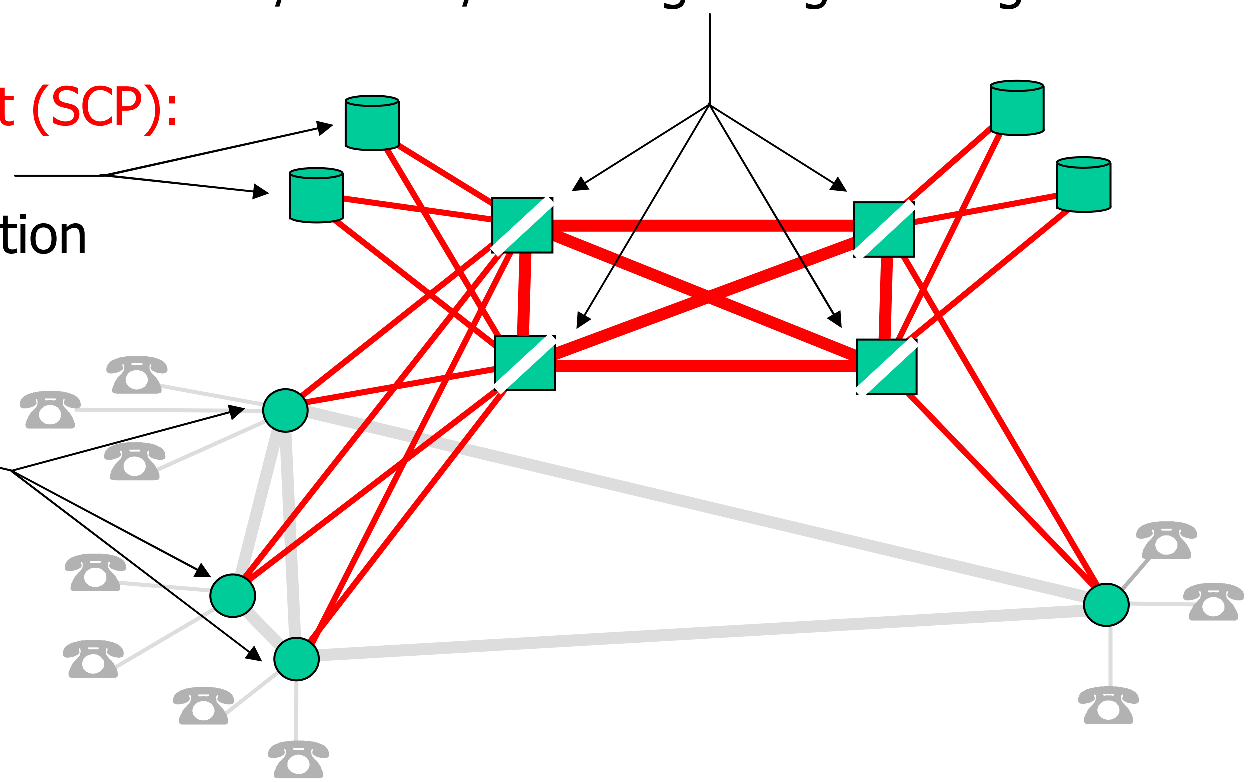
SS7: Telephone network

- Signaling *between telephone network elements:*

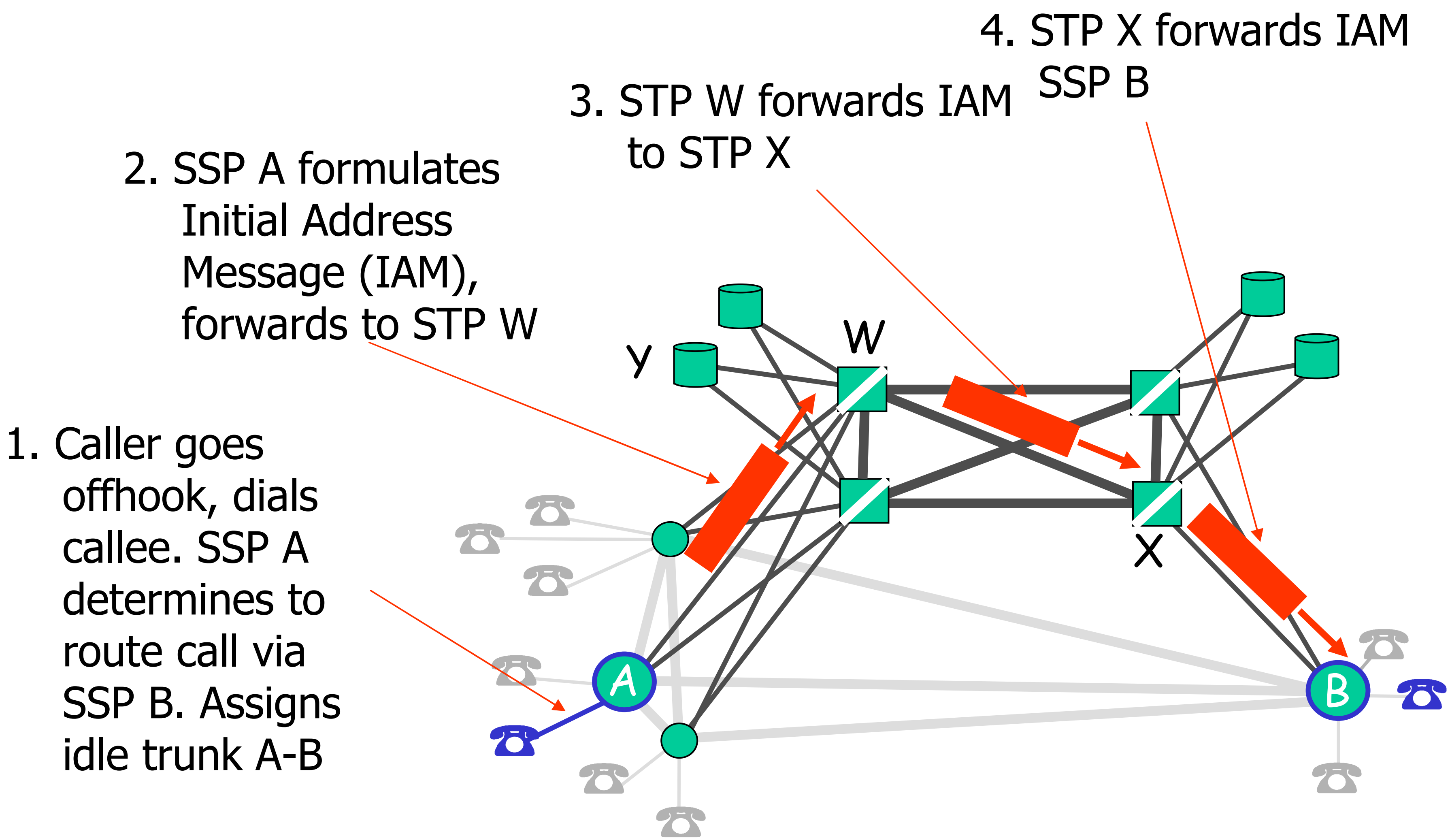
Signaling transfer point (STP):
- packet-switches of SS7 network
- send/receive/route signaling messages

Signaling control point (SCP):
- "Services" go here
- E.g., database function

Signaling switching point (SSP):
- Attach directly to end user
- Endpoints of SS7 network



Example: Signaling a POTS call

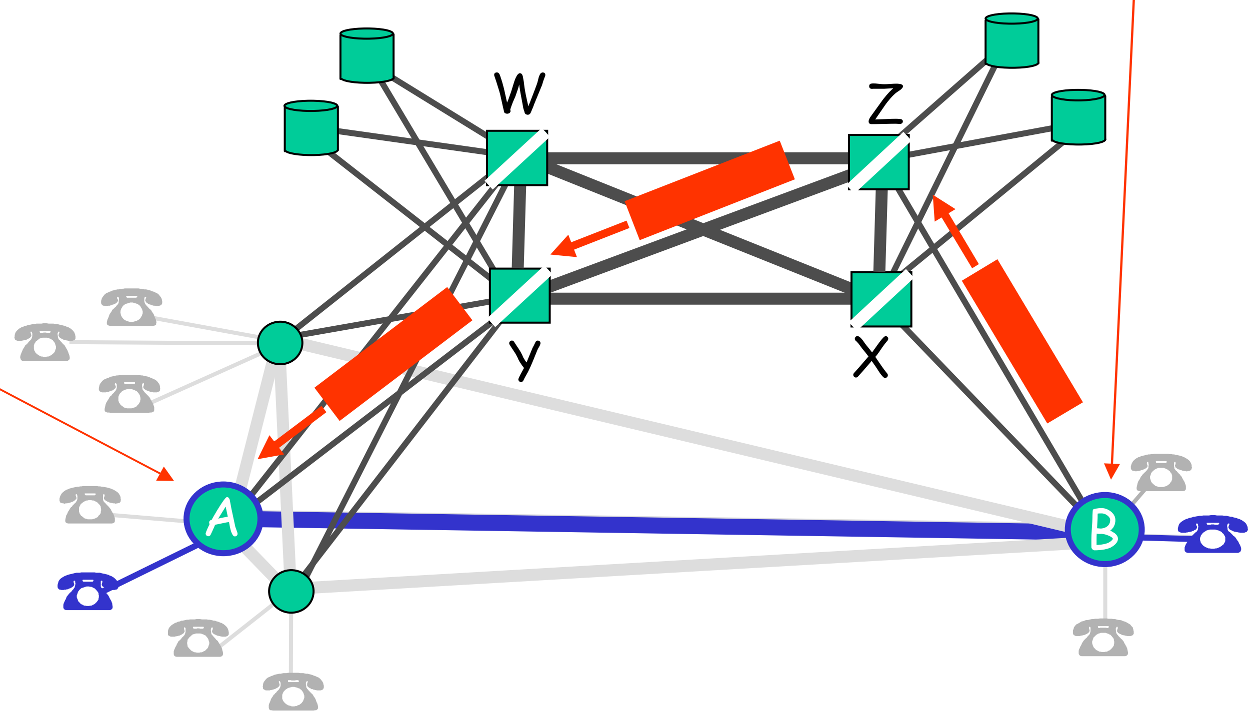


Example: Signaling a POTS call (2)

5. B determines it serves callee, creates address completion message (ACM[A,B,trunk]), rings callee phone, sends ringing sound on trunk to A

6. ACM routed to Z to Y to A

7. SSP A receives ACM, connects subscriber line to allocated A-B trunk (caller hears ringing)

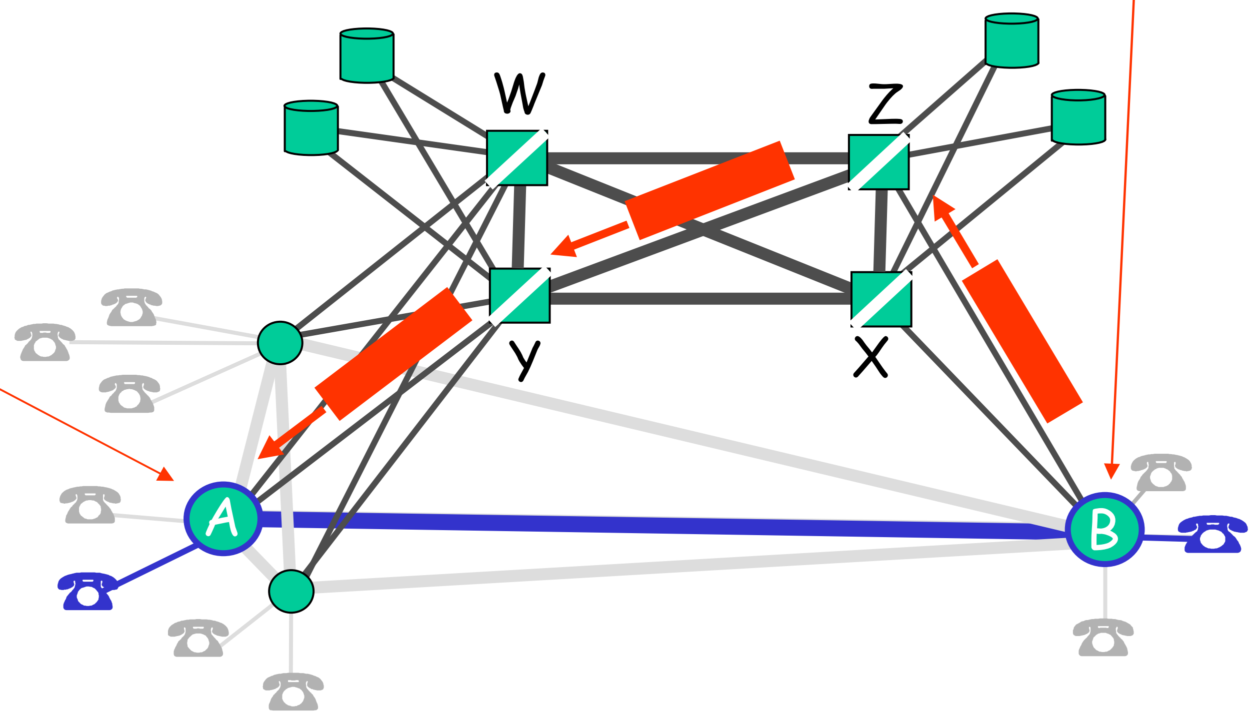


Example: Signaling a POTS call (3)

8. Callee goes off hook, B creates, sends answer message to A (ANM[A,B,trunk])

9. ANM routed to A

10. SSP A receives ANM, checks caller is connected in both directions to trunk.
Call is connected!



Example: Signaling a 800 call

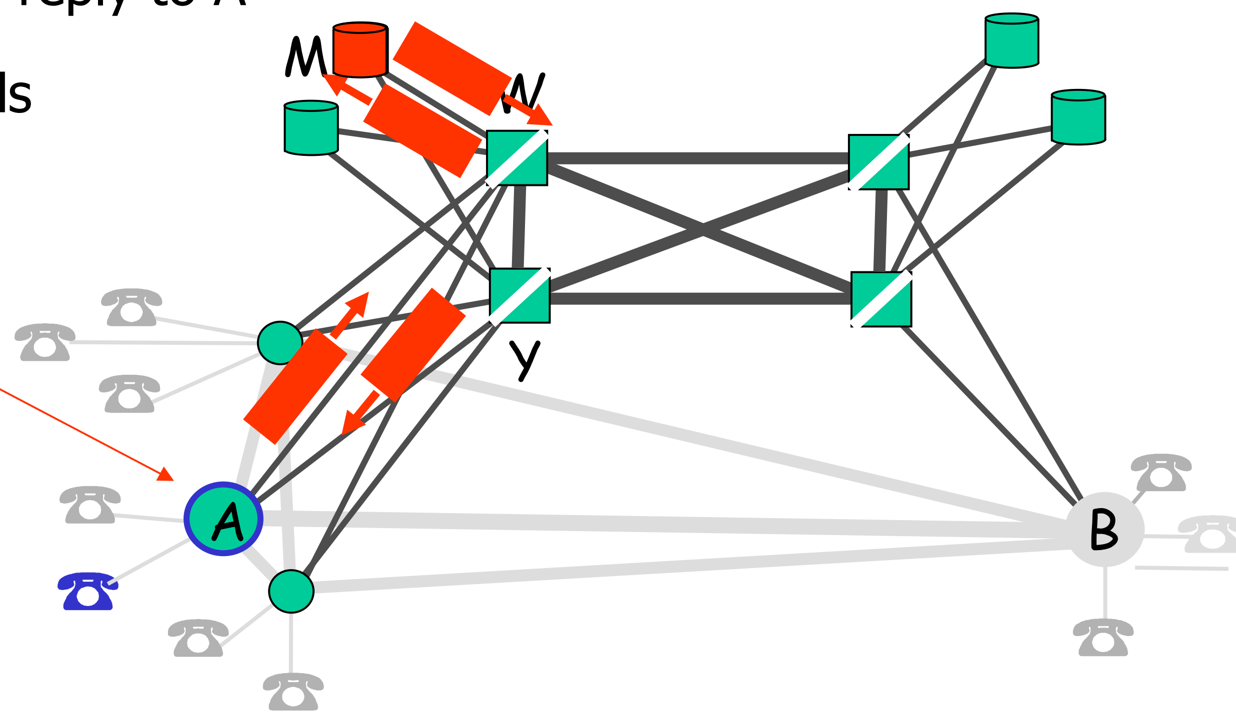
800 number: Logical phone number

- Translation to physical phone number needed, e.g., 1-800-CALL_ATT translates to 162-962-1943

3. M performs lookup, sends reply to A

2. STP W forwards request to M

1. Caller dials 800 number, A recognizes 800 number, formulates translation query, send to STP W

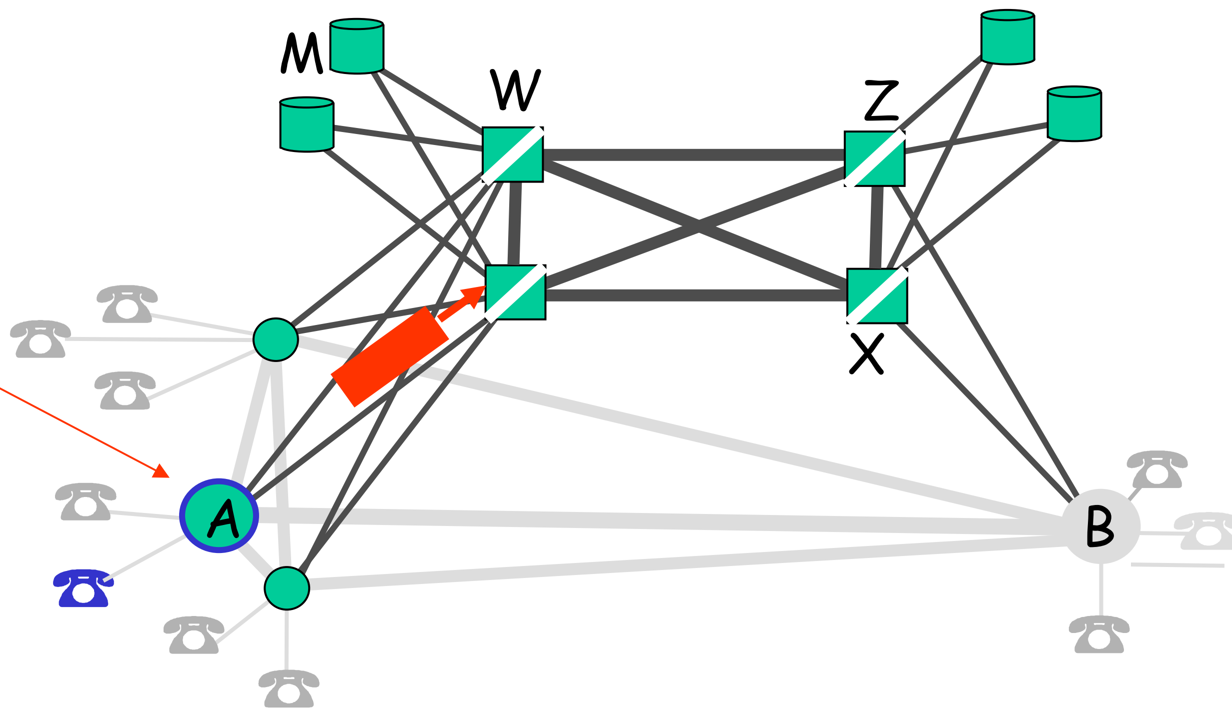


Example: Signaling a 800 call (2)

800 number: Logical phone number

- Translation to physical phone number needed

1. A begins signaling to set up call to number associated with 800 number



Example: SS7 protocol stack

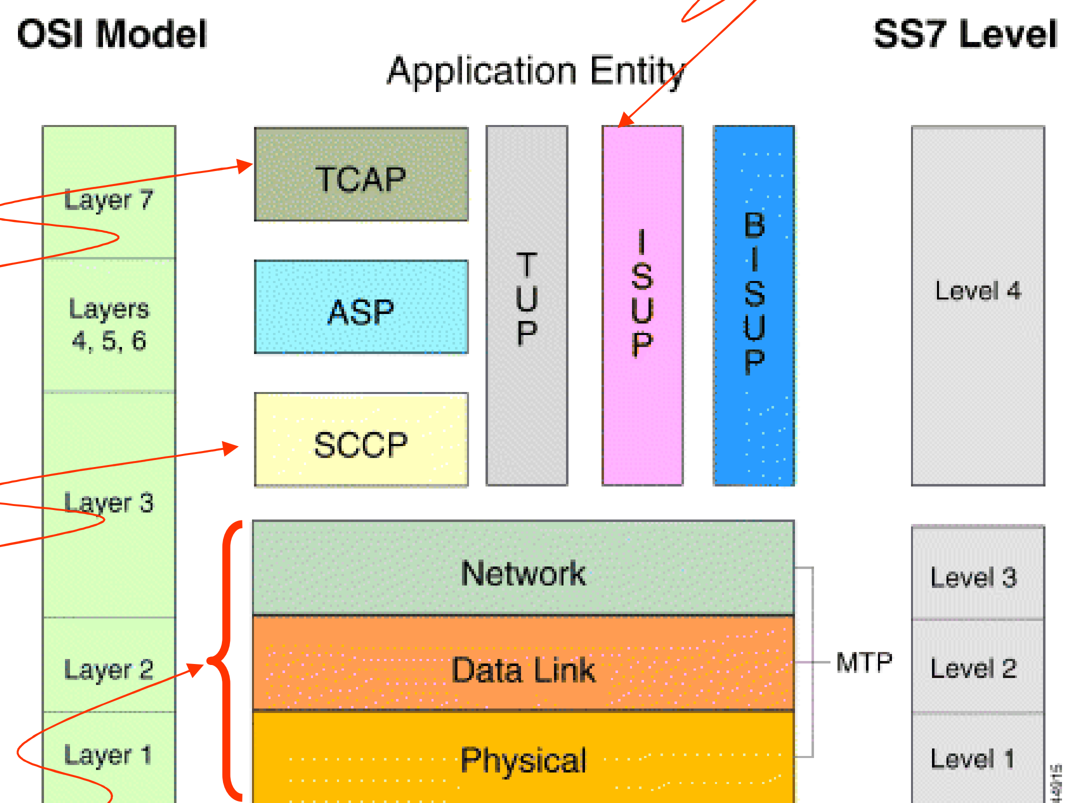
ISDN end-user signaling

TCAP: Application layer protocols: 800 service, calling card, call return, cellular roaming

SCCP: Demultiplexing to multiple upper layer applications

SS7-specific network, link, physical layer protocols

- move to IP (RFC 2719)?



TCAP: Transaction Capabilities Application Part
ASP: Application Service Part
SCCP: Signaling Connection Control Part
TUP: Telephone User Part
ISUP: ISDN User Part
BISUP: Broadband ISDN User Part
MTP: Message Transfer Part

Signaling: Discussion

- 800 logical-number-to-physical number translations: Looks like DNS
- Q: Differences:
  - In DNS end system generates request; dns is transparent to IP network- network layer in phone net does 800-service location translation for you [phone net has more "smarts in net]
  - DNS is more decentralized: hierarchical
  - Anyone can run a names server, but not a 800 server
  - ...
- Q: Where is state stored?
  - In POTS call: SSP stores circuit allocation, start/stop time
  - In 800 call: STPs know where to go for 800 service; in Internet, DNS location transparent to IP routers (knowledge of where to go to DNS service is in end-systems, not in router – intelligence at the edge)
- Q: Internet versus SS7/telephone network for accessing services

Signaling in the Internet

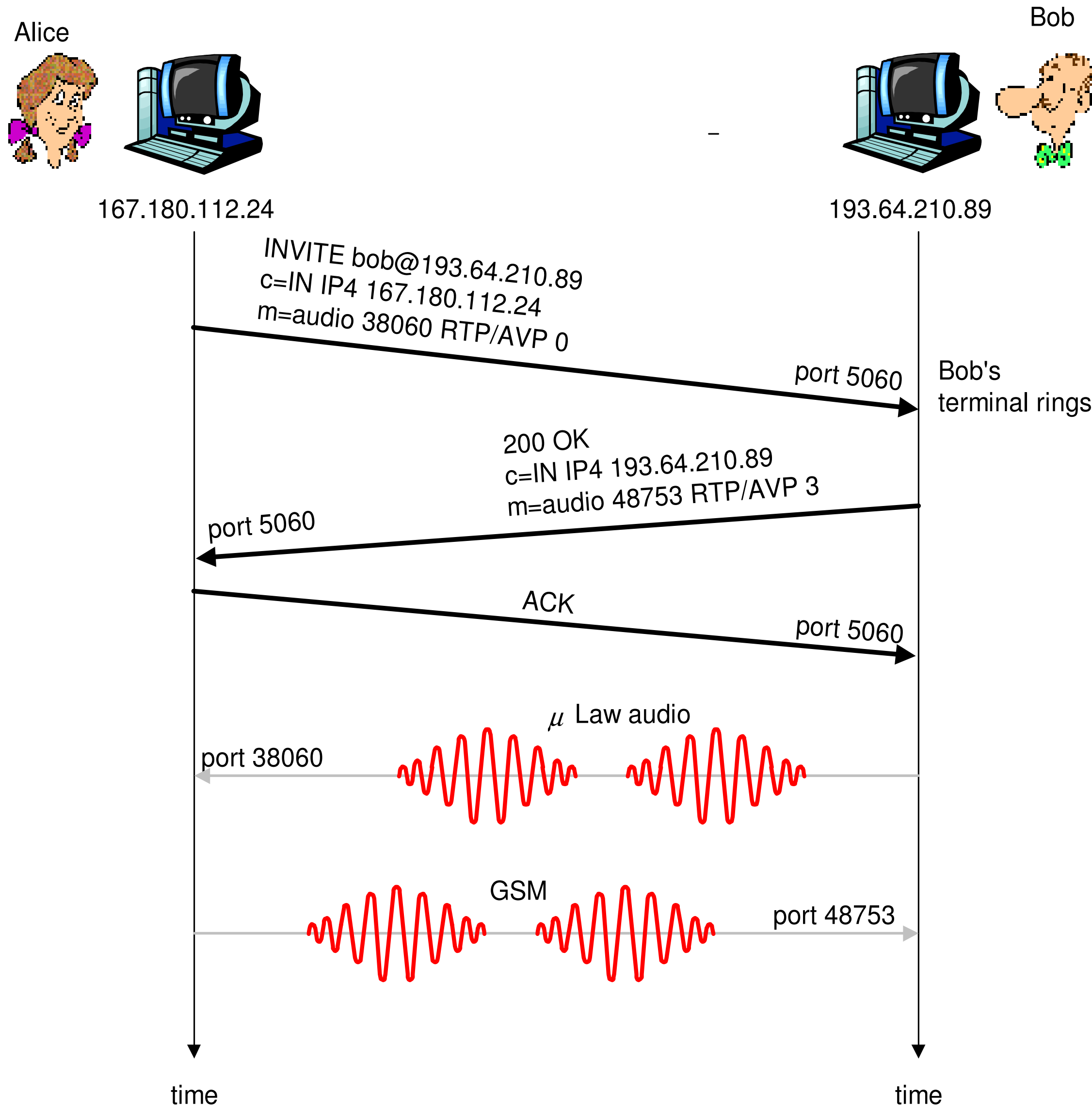
connectionless (stateless) forwarding by IP routers + best effort service = no network signaling protocols in initial IP design

- New requirement: Application layer protocol, that enables users to be reachable independent of the device and his location
- SIP: Session Initiation Protocol [RFC 3261]
  - IETF protocol
  - All telephone calls and video conference calls take place over the Internet
  - People are identified by names or e-mail addresses, rather than by phone numbers.
  - You can reach the callee, no matter where the callee roams, no matter what IP device the callee is currently using.

SIP Services

- Setting up a call
  - Provides mechanisms for caller to let callee know she wants to establish a call
  - Provides mechanisms so that caller and callee can agree on media type and encoding
  - Provides mechanisms to end call
- Determine current IP address of callee
  - Maps mnemonic identifier to current IP address
- Call management
  - Add new media streams during call
  - Change encoding during call
  - Invite others
  - Transfer and hold calls

Setting up a call to a known IP address



- Alice's SIP invite message indicates her port number & IP address. Indicates encoding that Alice prefers to receive (PCM ulaw)
- Bob's 200 OK message indicates his port number, IP address & preferred encoding (GSM)
- SIP messages can be sent over TCP or UDP; here sent over RTP/UDP
- Default SIP port number is 5060

Setting up a call (more)

- Codec negotiation
  - Suppose Bob doesn't have PCM ulaw encoder
  - Bob will instead reply with 606 Not Acceptable Reply and list encoders he can use
  - Alice can then send a new INVITE message, advertising an appropriate encoder
- Rejecting a call
  - Bob can reject with replies "busy," "gone," "payment required," "forbidden"
- Media can be sent over RTP or some other protocol

Example of SIP message

```
INVITE sip:bob@domain.com SIP/2.0
Via: SIP/2.0/UDP 167.180.112.24
From: sip:alice@hereway.com
To: sip:bob@domain.com
Call-ID: a2e3a@pigeon.hereway.com
Content-Type: application/sdp
Content-Length: 885

c=IN IP4 167.180.112.24
m=audio 38060 RTP/AVP 0
```

Notes:

- HTTP message syntax
- Sdp = session description protocol
- Call-ID is unique for every call.

- Here we don't know Bob's IP address. Intermediate SIP servers is necessary
- Alice sends and receives SIP messages using the SIP default port number 506
- Alice specifies in Via: Header that SIP client sends and receives SIP messages over UDP

Name translation and user location

- Caller wants to call callee, but only has callee's name or e-mail address.
- Need to get IP address of callee's current host:
  - User moves around
  - DHCP protocol
  - User has different IP devices (PC, PDA, car device)
- Result can be based on:
  - Time of day (work, home)
  - Caller (don't want boss to call you at home)
  - Status of callee (calls sent to voicemail when callee is already talking to someone)

Service provided by SIP servers:

- SIP registrar server
- SIP proxy server

SIP registrar

- When Bob starts SIP client, client sends SIP REGISTER message to Bob's registrar server
  (similar function needed by Instant Messaging)

Register Message:

```
REGISTER sip:domain.com SIP/2.0
Via: SIP/2.0/UDP 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@domain.com
Expires: 3600
```

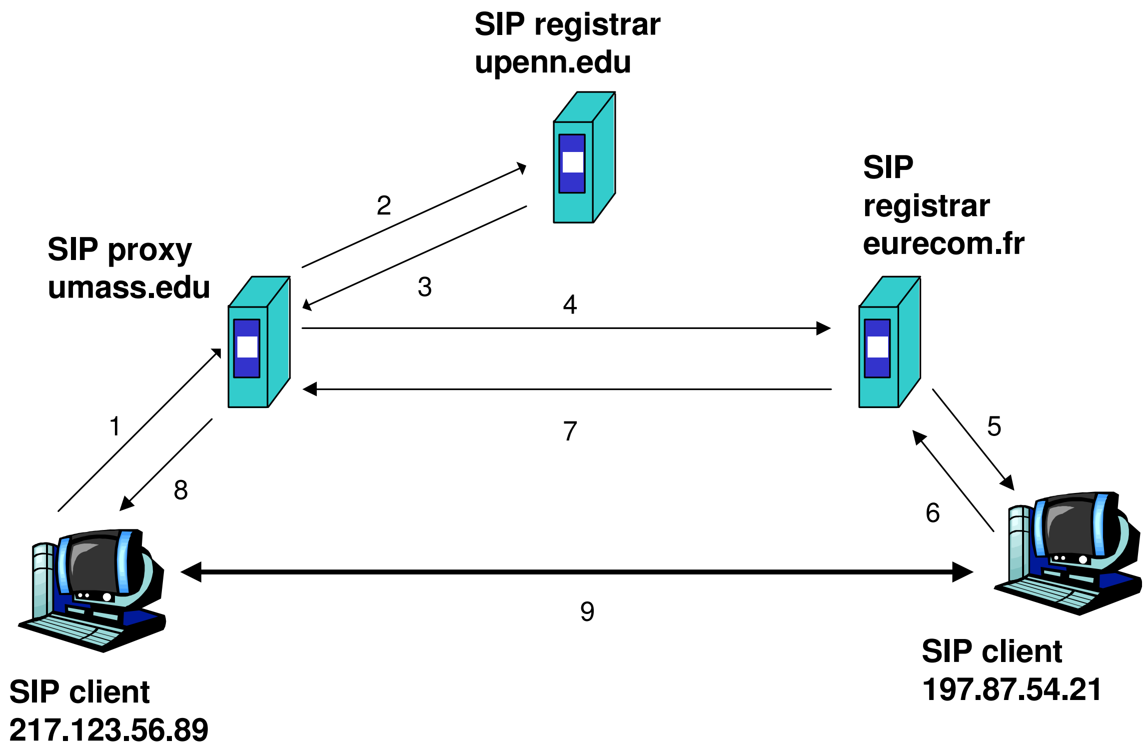
SIP proxy

- Alice send’s invite message to her proxy server
  - Contains address sip:bob@domain.com
- Proxy responsible for routing SIP messages to callee
  - Possibly through multiple proxies.
- Callee sends response back through the same set of proxies.
- Proxy returns SIP response message to Alice
  - Contains Bob’s IP address
- Note: proxy is analogous to local DNS server

Example

Caller jim@umass.edu
with places a
call to keith@upenn.edu

(1) Jim sends INVITE message to umass SIP proxy. (2) Proxy forwards request to upenn registrar server.
(3) Upenn server returns redirect response, indicating that it should try keith@eurecom.fr



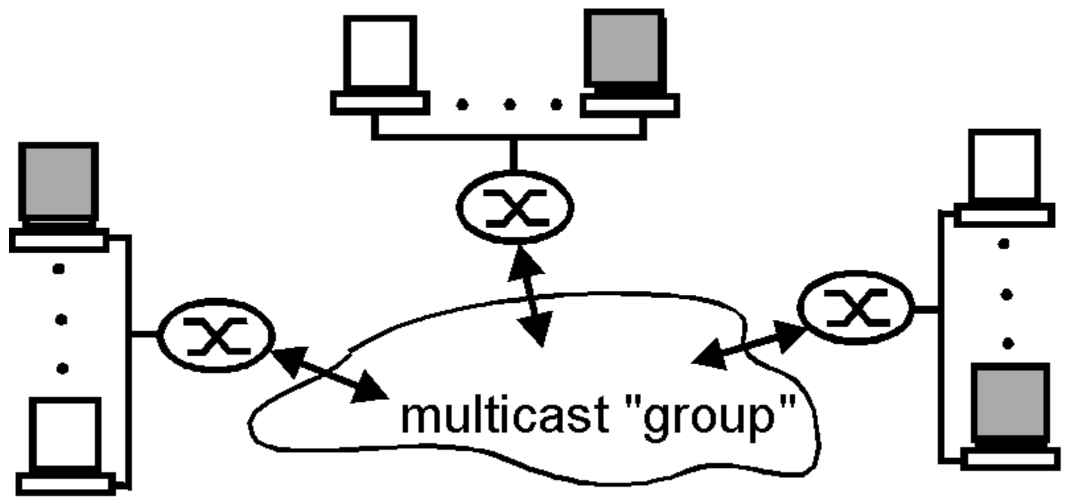
(4) Umass proxy sends INVITE to eurecom registrar. (5) eurecom regristrar forwards INVITE to 197.87.54.21, which is running keith's SIP client. (6-8) SIP response sent back (9) media sent directly
between clients.
Note: Also a SIP ack message, which is not shown.

Signaling in the Internet

connectionless (stateless) forwarding by IP routers + best effort service = no network signaling protocols in initial IP design

- New requirement: reserve resources along end-to-end path (end system, routers) for QoS for multimedia applications
- RSVP: Resource Reservation Protocol [RFC 2205]
  - " … allow users to communicate requirements to network in robust and efficient way." i.e., signaling !
  - Earlier Internet Signaling protocol: ST-II [RFC 1819]
  - Designed with multicast in mind

Internet multicast service model



Multicast group concept:

- Hosts send IP datagram pkts to multicast group
- Hosts that have “joined” that multicast group will receive pkts sent to that group
- Routers forward multicast datagrams to hosts

Multicast groups

- Class D Internet addresses reserved for multicast:

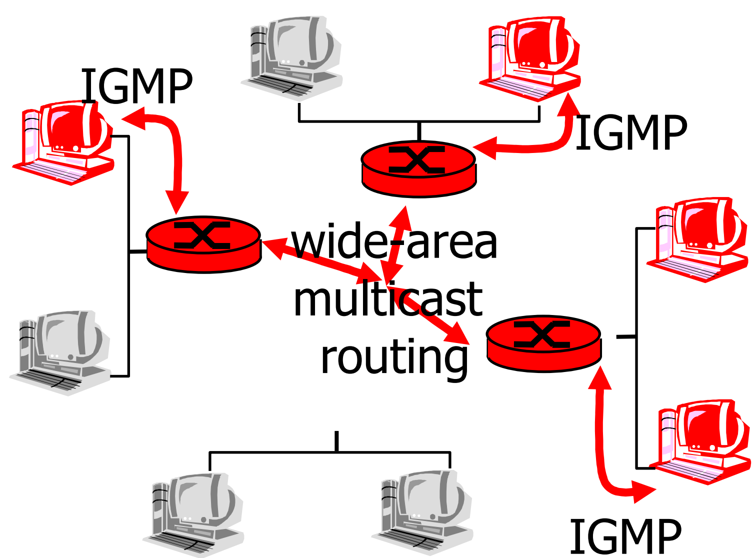
1110	Multicast Group ID

← 28 bits →

- Host group semantics:
  - o Anyone can "join" (receive) multicast group
  - o Anyone can send to multicast group
  - o No network-layer identification to hosts of members
- *Needs:* Infrastructure to deliver mcast-addressed datagrams to all hosts that have joined that multicast group

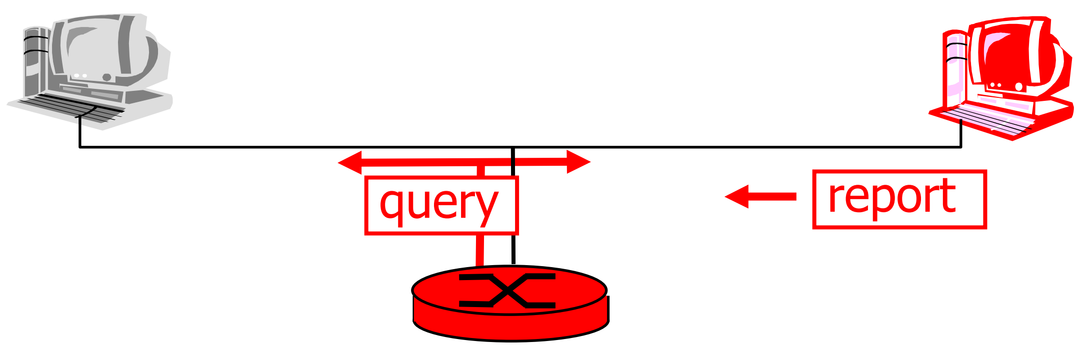
Joining a mcast group: Two-step process

- *Local:* Host informs local mcast router of desire to join group: IGMP (Internet Group Management Protocol)
- *Wide area:* Local router interacts with other routers to receive mcast datagram flow
    - Many protocols (e.g., DVMRP, MOSPF, PIM)



IGMP: Internet Group Management Protocol

- *Host:* sends IGMP report when application joins mcast group
  - IP_ADD_MEMBERSHIP socket option
  - Host need not explicitly "unjoin" group when leaving
- *Router:* sends IGMP query at regular intervals
  - Host belonging to a mcast group must reply to query



IGMP

IGMP version 1

- *Router:* Host Membership Query msg broadcast on LAN to all hosts
- *Host:* Host Membership Report msg to indicate group membership
  - Randomized delay before responding
  - Implicit leave via no reply to Query
- RFC 1112

IGMP v2: additions include

- Group-specific Query
- Leave Group msg
  - Last host replying to Query can send explicit Leave Group msg
  - Router performs group-specific query to see if any hosts left in group
  - RFC 2236

IGMP v3: Internet draft

RSVP design goals

1. Accommodate heterogeneous receivers (different bandwidth along paths)
2. Accommodate different applications with different resource requirements
3. Make multicast a first class service, with adaptation to multicast group membership
4. Leverage existing multicast/unicast routing, with adaptation to changes in underlying unicast, multicast routes
5. Control protocol overhead to grow (at worst) linear in # receivers
6. Modular design for heterogeneous underlying technologies

RSVP does not…

- Specify how resources are to be reserved
  - Rather: a mechanism for communicating needs
- Determine routes packets will take
  - That's the job of routing protocols
  - Signaling decoupled from routing
- Interact with forwarding of packets
  - Separation of control (signaling) and data (forwarding) planes

RSVP: Overview of operation

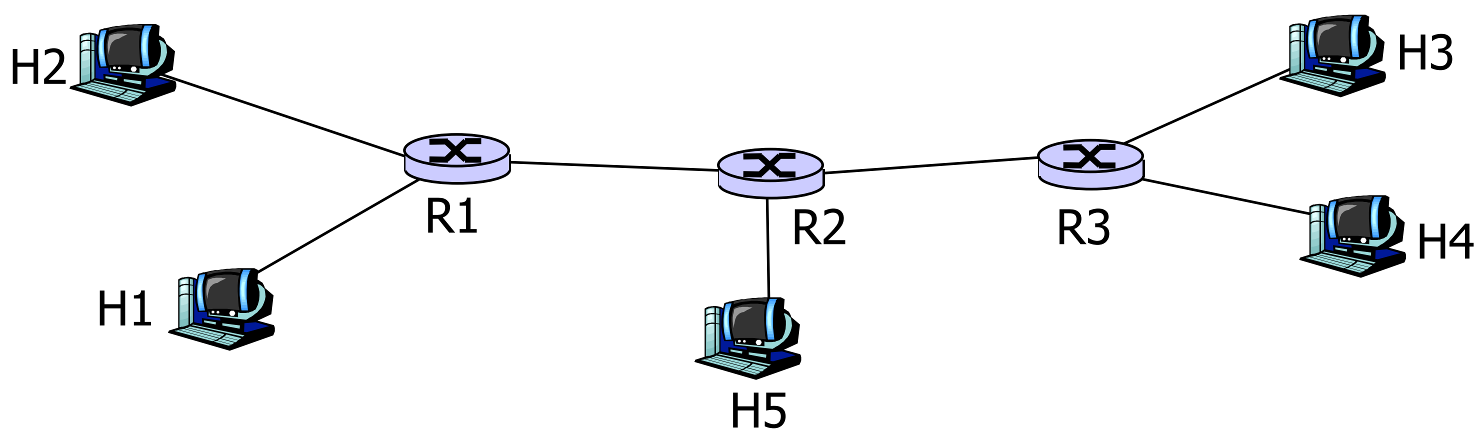
- Senders, receiver join a multicast group
  - Done outside of RSVP
  - Senders need not join group
- Sender-to-network signaling
  - *Path message:* make sender presence known to routers
  - Path teardown: delete sender's path state from routers
- Receiver-to-network signaling
  - *Reservation message:* reserve resources from sender(s) to receiver
  - Reservation teardown: remove receiver reservations
- Network-to-end-system signaling
  - Path error
  - Reservation error

Path msgs: RSVP *sender-to-network* signaling

- Path message contents:
  - *Address:* unicast destination, or multicast group
  - *Flowspec:* bandwidth requirements spec.
  - *Filter flag:* if yes, record identities of upstream senders (to allow packets filtering by source)
  - *Previous hop:* upstream router/host ID
  - *Refresh time:* time until this info times out
- Path message: communicates sender info, and reverse-path-to-sender routing info
  - Later upstream forwarding of receiver reservations

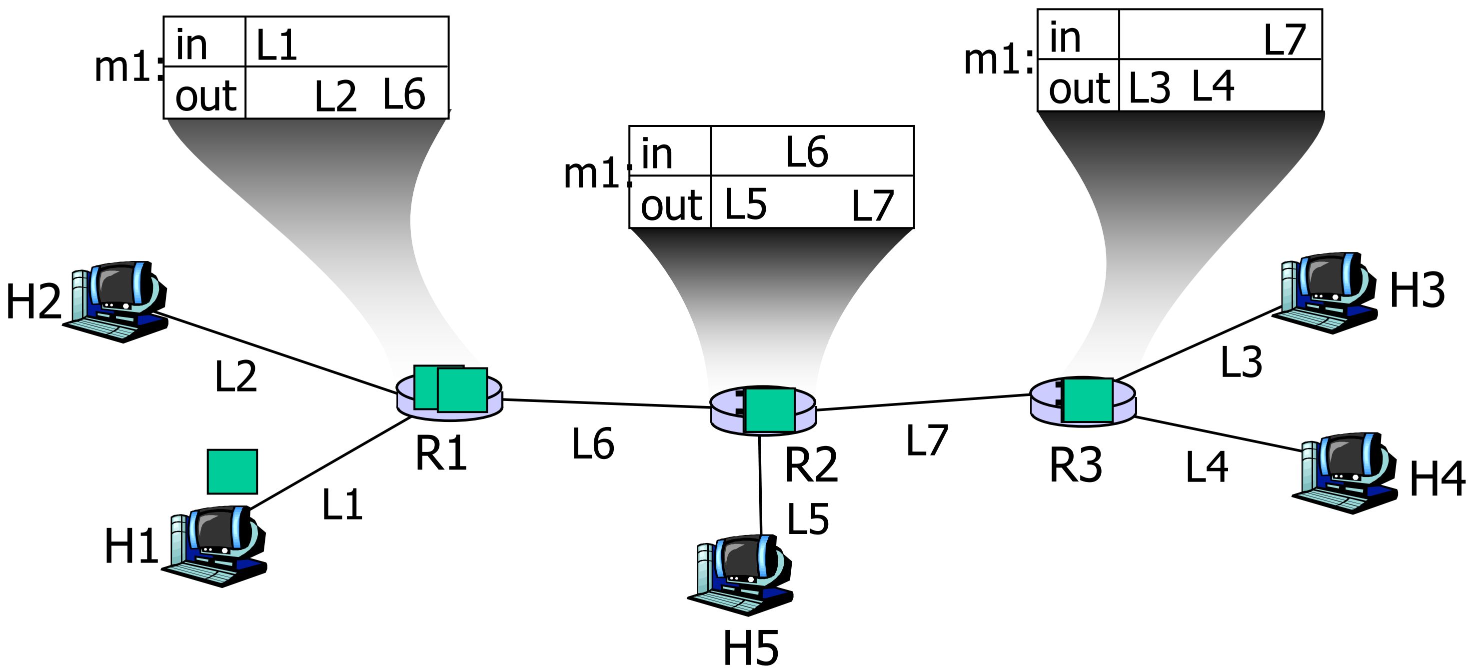
RSVP: Simple audio conference

- H1, H2, H3, H4, H5 both senders and receivers
- Multicast group m1
- No filtering: packets from any sender forwarded
- Audio rate: *b*
- Only one multicast routing tree possible



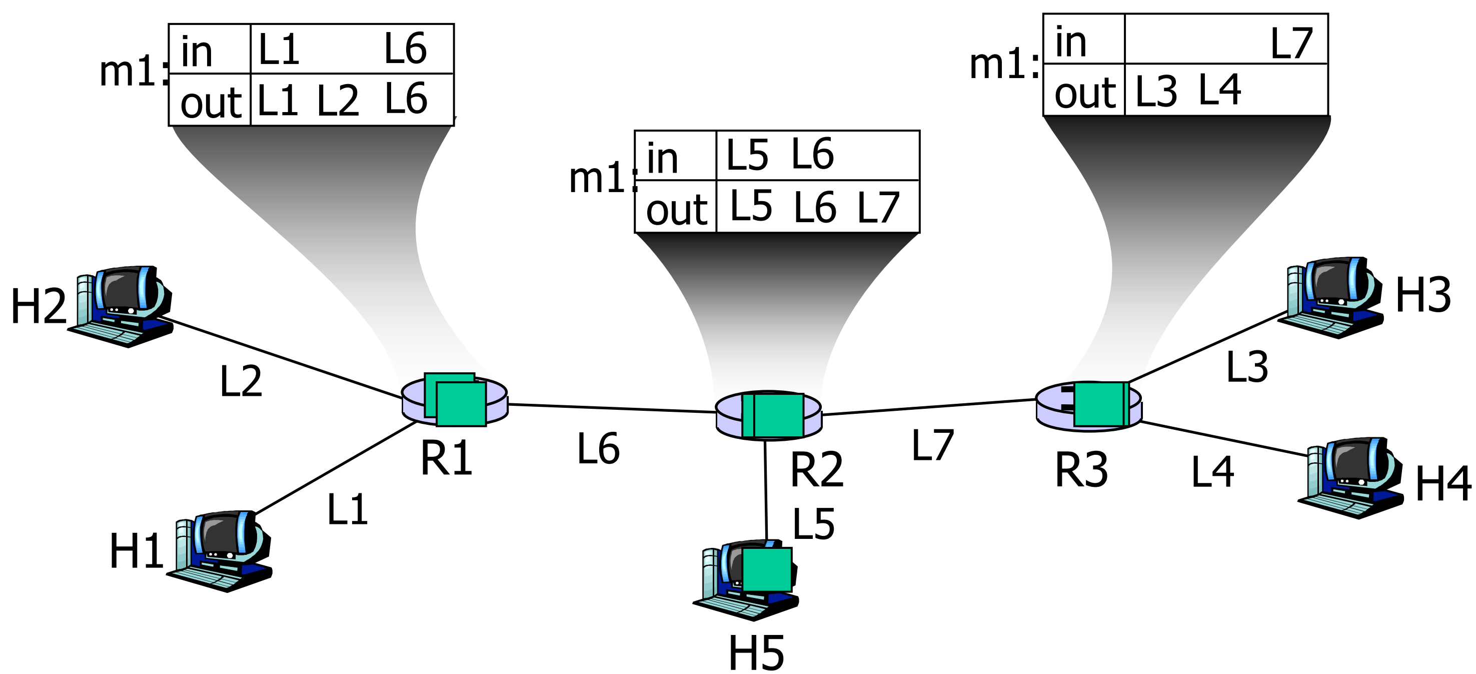
RSVP: Building up path state

- H1, …, H5 all send path messages on *m1:*
  (address=*m1*, Tspec=*b*, filter-spec=no-filter,refresh=100)
- Suppose H1 sends first path message



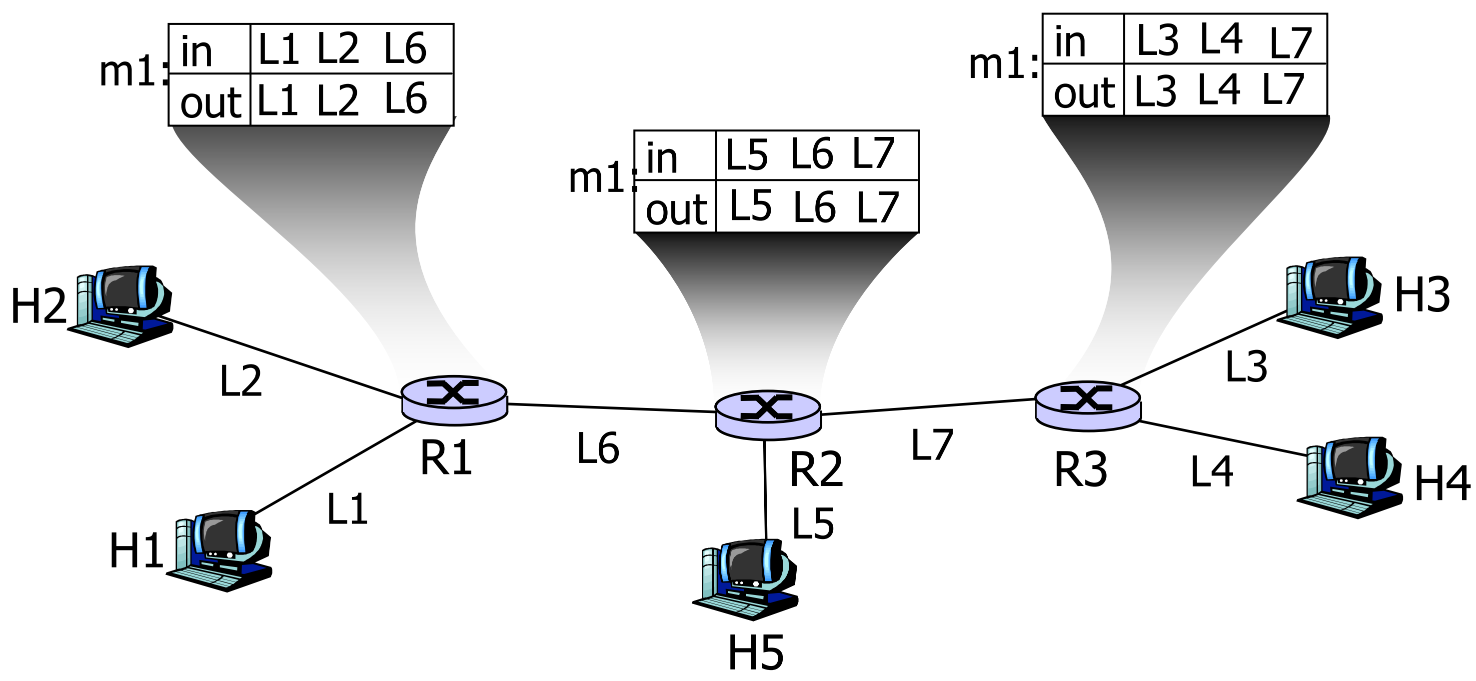
RSVP: Building up path state

- Next, H5 sends path message, creating more state in routers



RSVP: Building up path state

- H2, H3, H5 send path msgs, completing path state tables



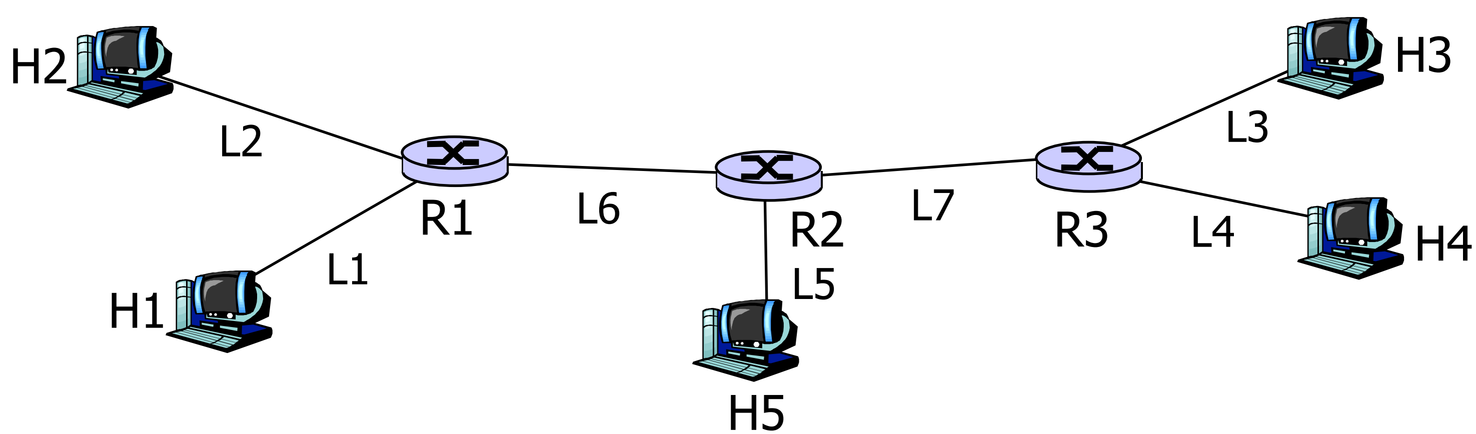
Reservation msgs: *Receiver-to-network* signaling

- Reservation message contents:
  - *Desired bandwidth:*
  - *Filter type:*
    - No filter: any packets address to multicast group can use reservation
    - Fixed filter: only packets from specific set of senders can use reservation
    - Dynamic filter: set of senders that can use reservation changes over time
  - *Filter spec*
- Reservations flow upstream from receiver-to-senders, reserving resources, creating additional, *receiver-related* state at routers

RSVP: *Receiver* reservation – example 1

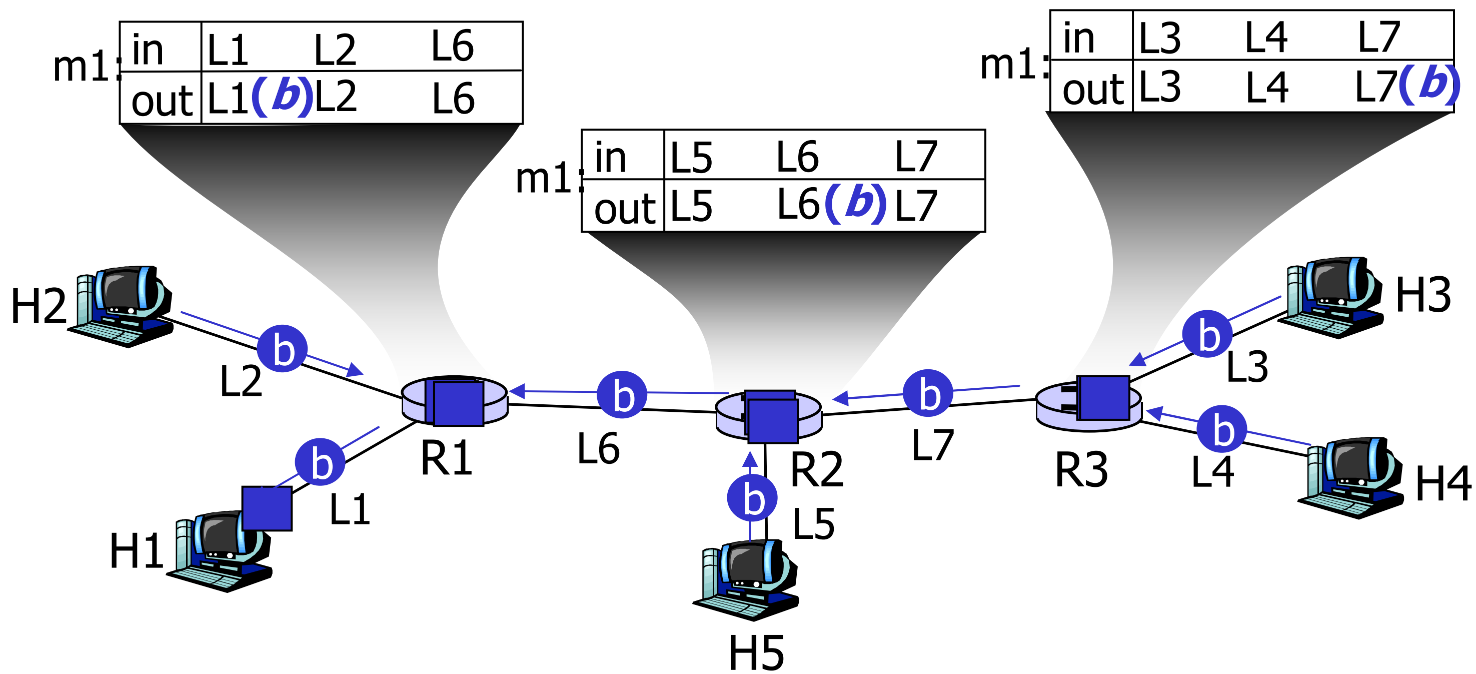
H1 wants to receive audio from all other senders

- H1 reservation msg flows uptree to sources
- H1 only reserves enough bandwidth for 1 audio stream
- Reservation is of type "no filter" – any sender can use reserved bandwidth



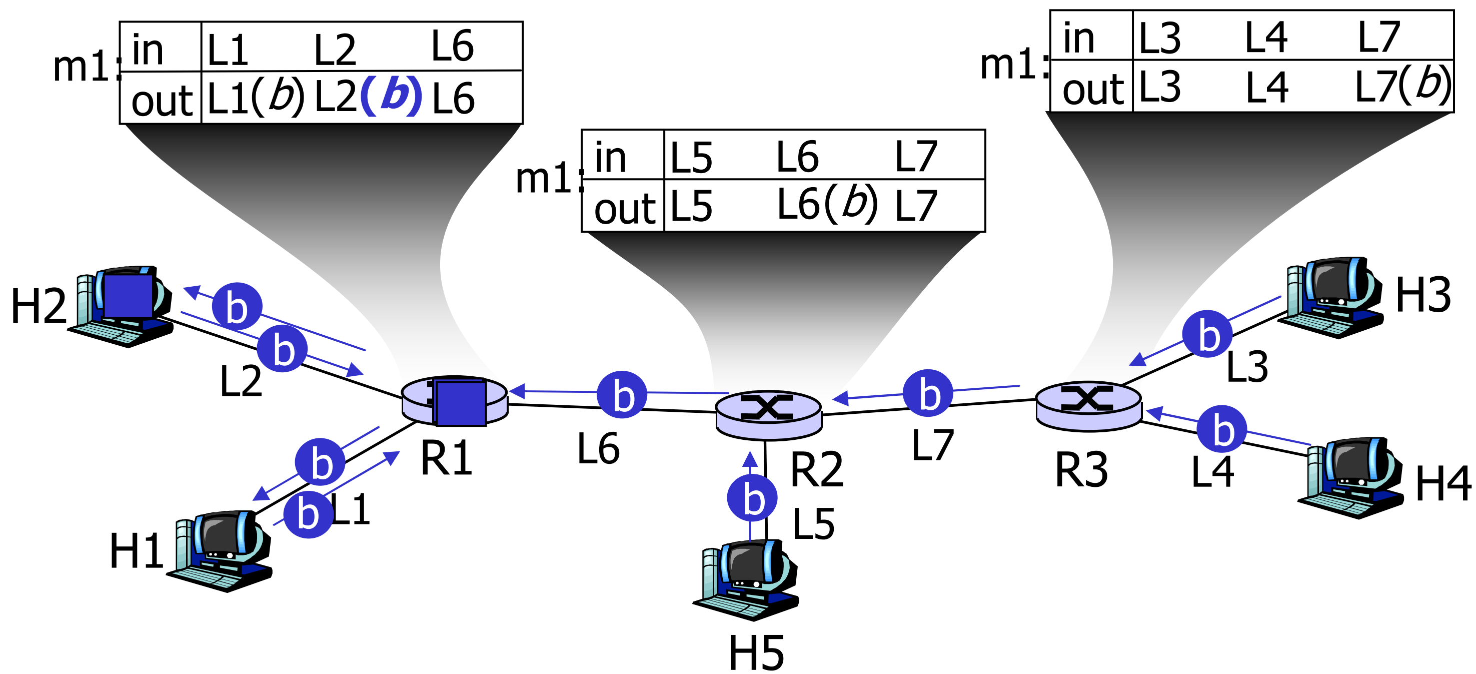
RSVP: *Receiver* reservation – example 1

- H1 reservation msgs flows uptree to sources
- Routers, hosts reserve bandwidth b needed on downstream links towards H1



RSVP: *Receiver* reservation – example 1

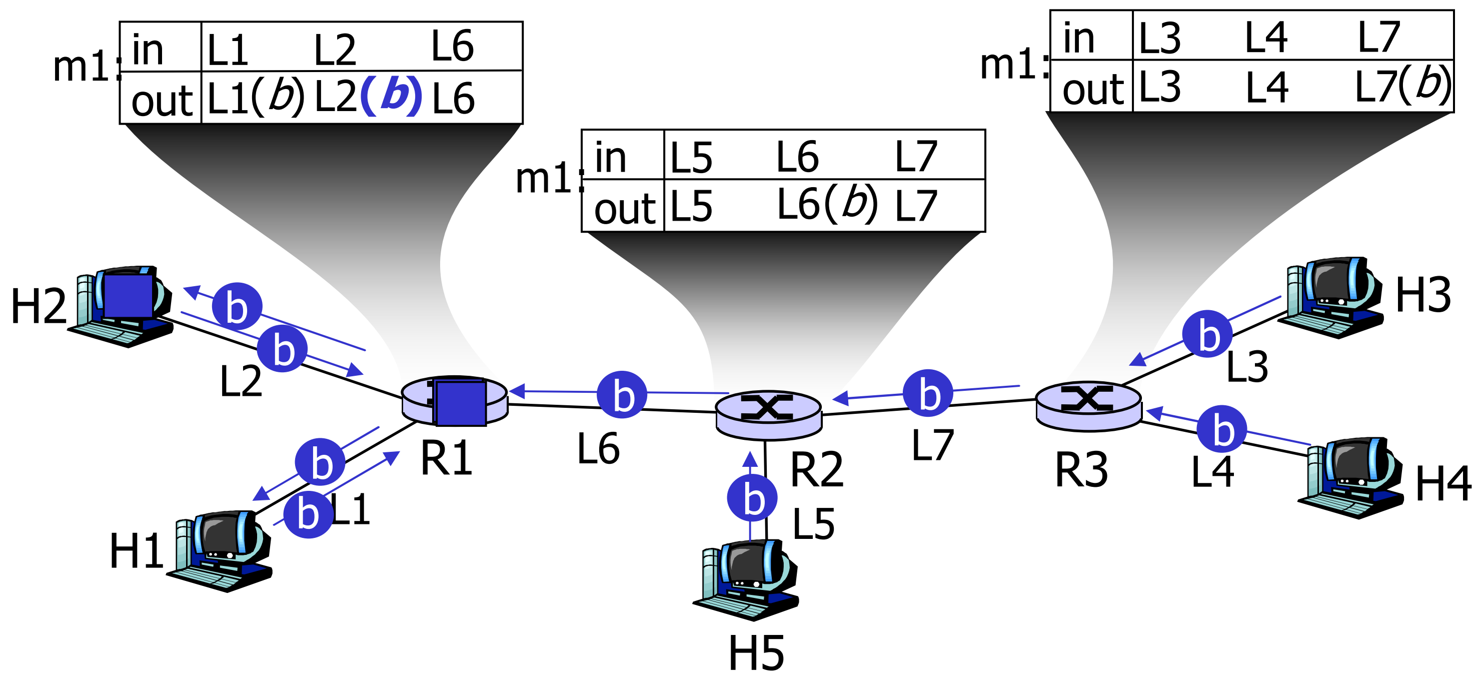
- next, H2 makes no-filter reservation for bandwidth *b*
- H2 forwards to R1, R1 forwards to H1 and R2 (?)
- R2 takes no action, since *b* already reserved on L6



RSVP: *Receiver* reservation – issues

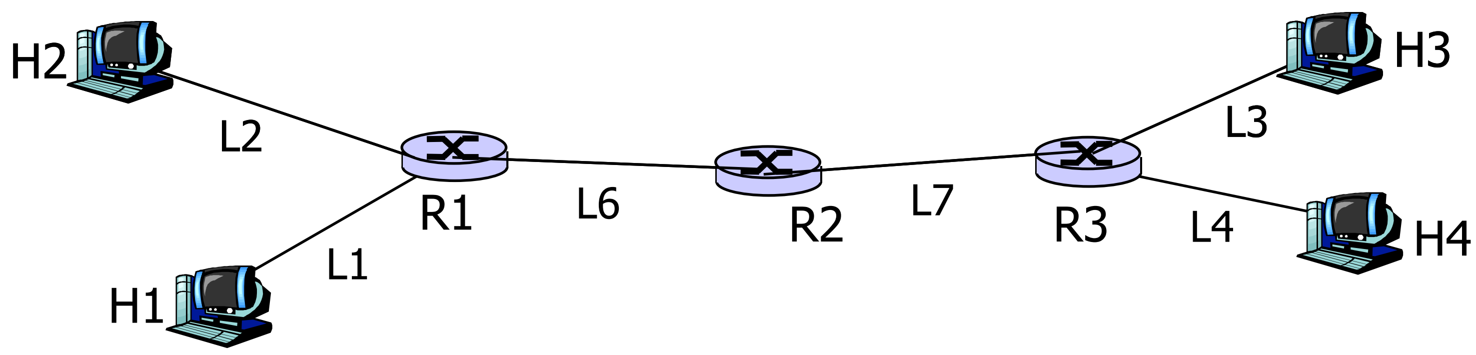
What if multiple senders (e.g., H3, H4, H5) over link (e.g., L6)?

- Arbitrary interleaving of packets
- L6 flow policed by leaky bucket: if H3+H4+H5 sending rate exceeds b, packet loss will occur



RSVP: Example 2

- H1, H4 are only senders
  - send *path messages* as before, indicating filtered reservation
  - Routers store upstream senders for each upstream link
- H2 will want to receive from H4 (only)



RSVP: Example 2

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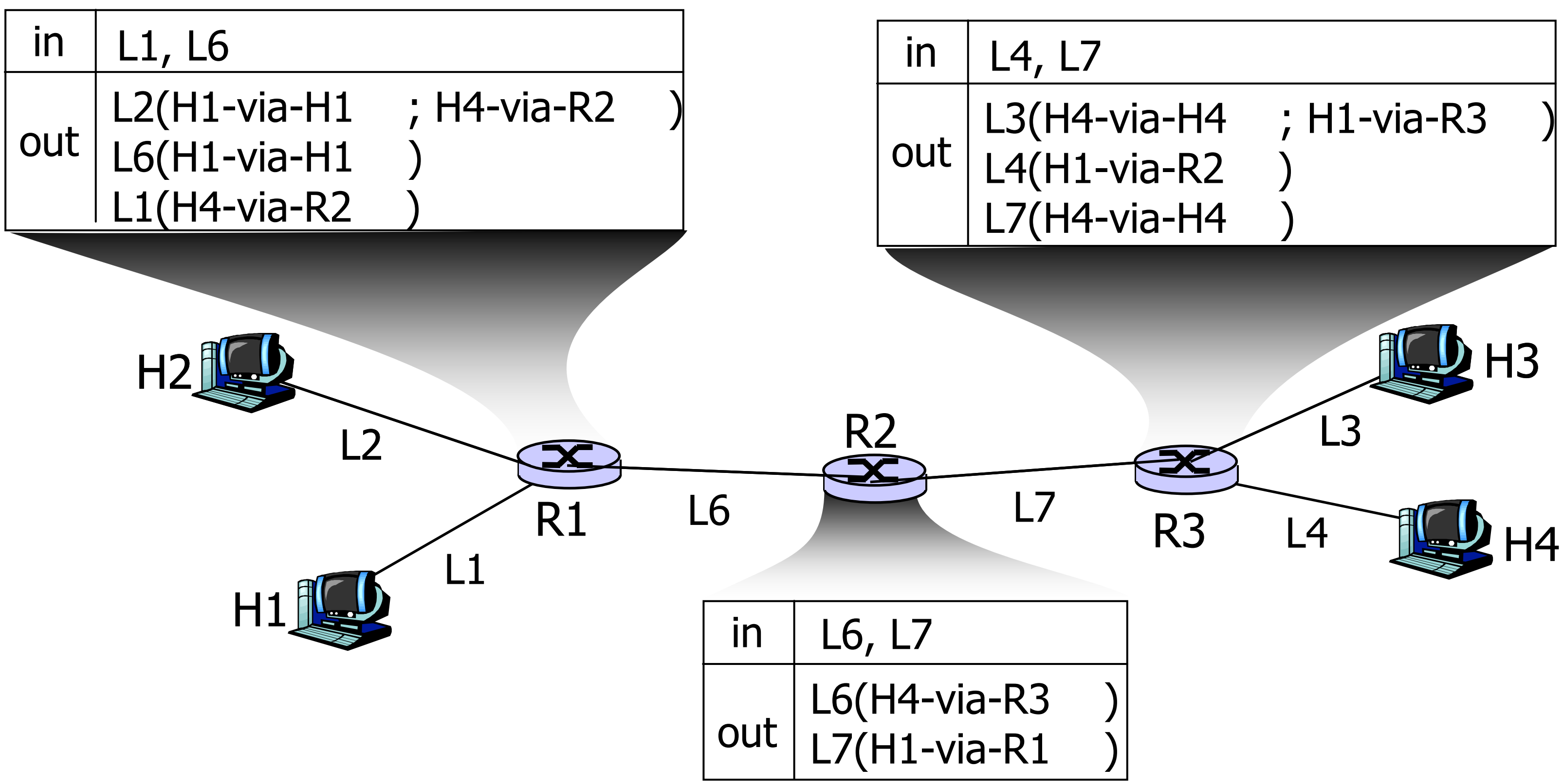
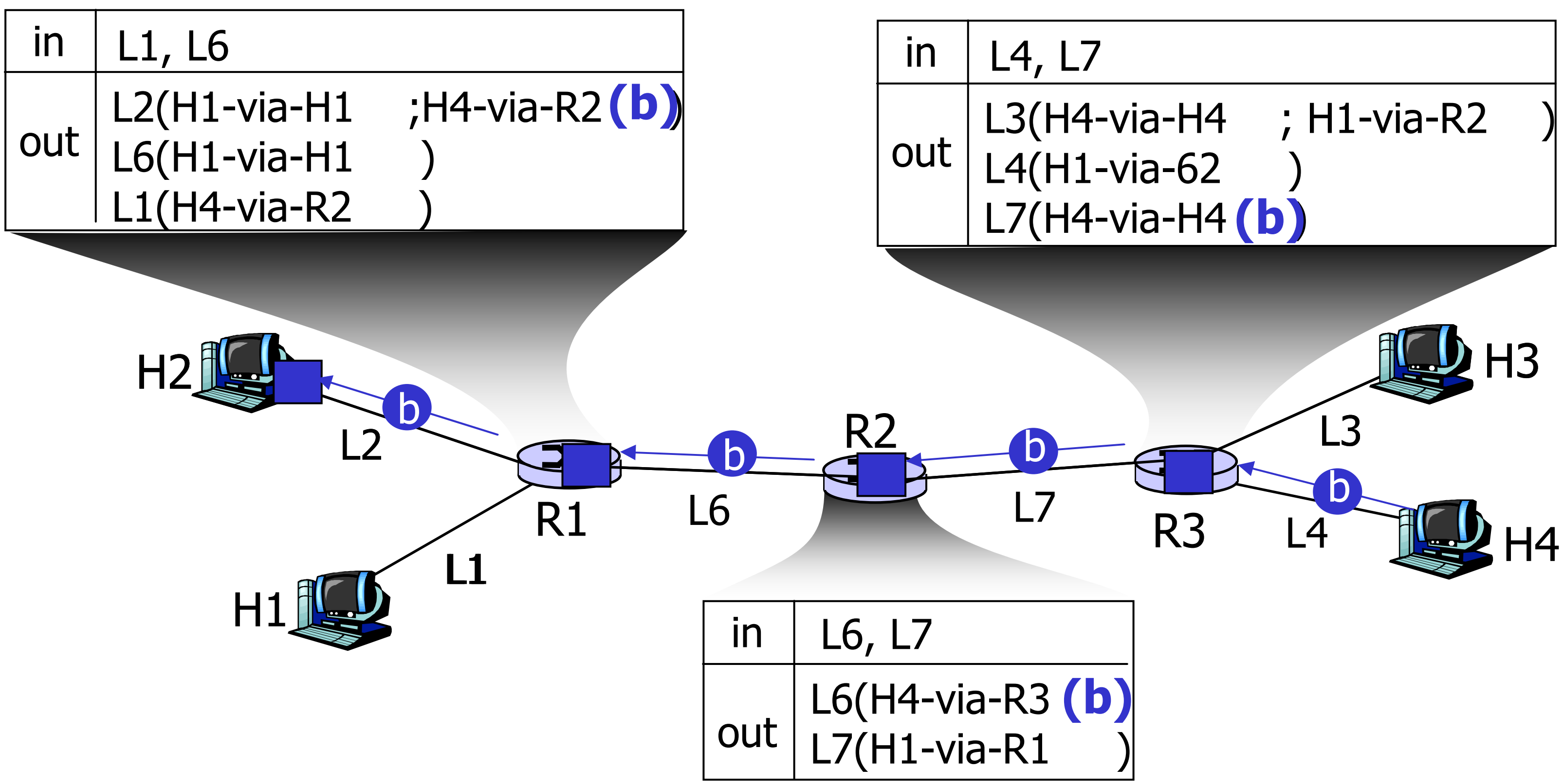
  - Send *path messages* as before, indicating filtered reservation



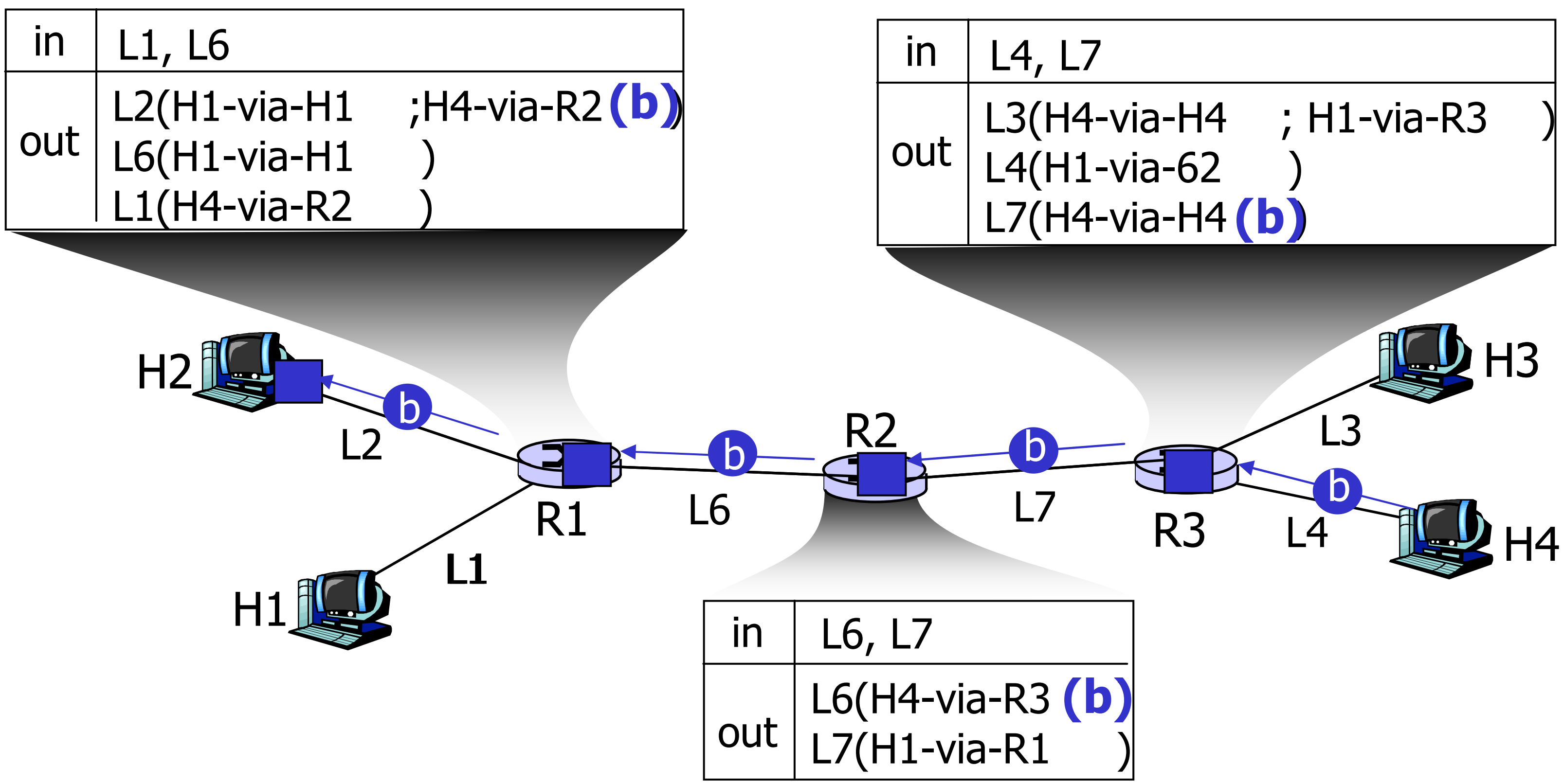
RSVP: Example 2

- Receiver H2 sends reservation message for source H4 at bandwidth *b*
  - Propagated upstream towards H4, reserving *b*



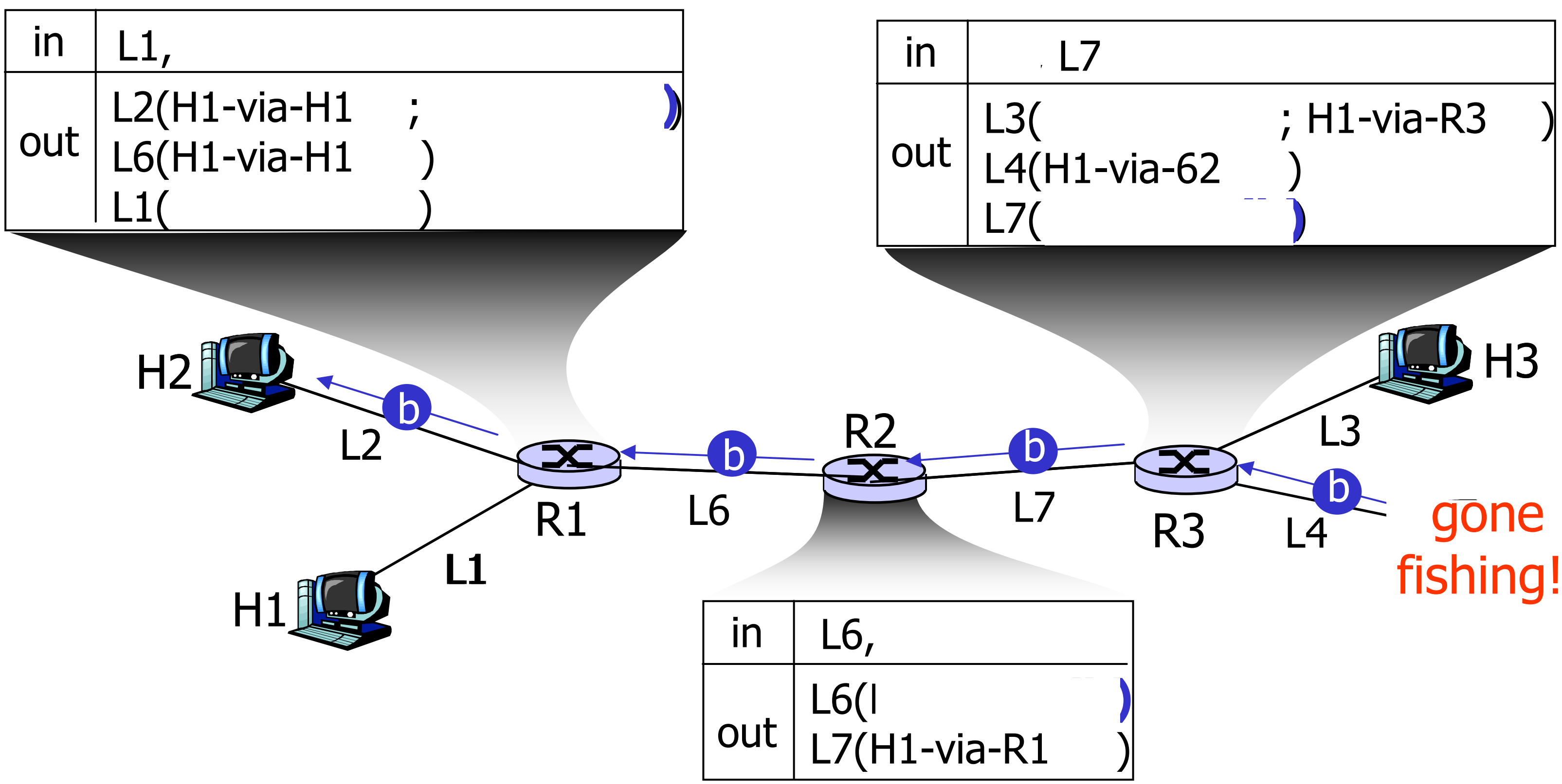
RSVP: *Soft-state*

- Senders periodically resend path msgs to refresh (maintain) state
- Receivers periodically resend resv msgs to refresh (maintain) state
- Path and resv msgs have TTL field, specifying refresh interval



RSVP: *Soft-state*

- Suppose H4 (sender) leaves without performing teardown
- Eventually state in routers will timeout and disappear!



Use cases for reservation/path refresh

- Recover from an earlier lost refresh message
  - Expected time until refresh received must be shorter than timeout interval! (short timer interval desired)
- Handle receiver/sender that goes away without teardown
  - Sender/receiver state will timeout and disappear
- Reservation refreshes will cause new reservations to be made to a receiver from a sender who has joined since receivers last reservation refresh
  - E.g., in previous example, H1 is only receiver, H3 only sender Path/reservation messages complete, data flows
  - H4 joins as sender, nothing happens until H3 refreshes reservation, causing R3 to forward reservation to H4, which allocates bandwidth

RSVP: reflections

- Multicast as a “first class” service
- Receiver-oriented reservations
- Use of soft-state

Signaling Discussion:

SS7 vs. TCP vs. SIP vs. RSVP

- Similarities
  -
  -
  -
  -
- Differences
  -
  -
  -