COMPUTER NETWORKS
CS 45201
CS 55201

CHAPTER 5
End-to-End protocols

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Contents

End-to-End (Transport) Protocols

- Underlying best-effort network
  - drops messages
  - re-orders messages
  - delivers duplicate copies of a given message
  - limits messages to some finite size
  - delivers messages after an arbitrarily long delay
- Common end-to-end services
  - guarantee message delivery
  - deliver messages in the same order they are sent
  - deliver at most one copy of each message
  - support arbitrarily large messages
  - support synchronization
  - allow the receiver to apply flow control to the sender
  - support multiple application processes on each host

Simple Demultiplexer (UDP)

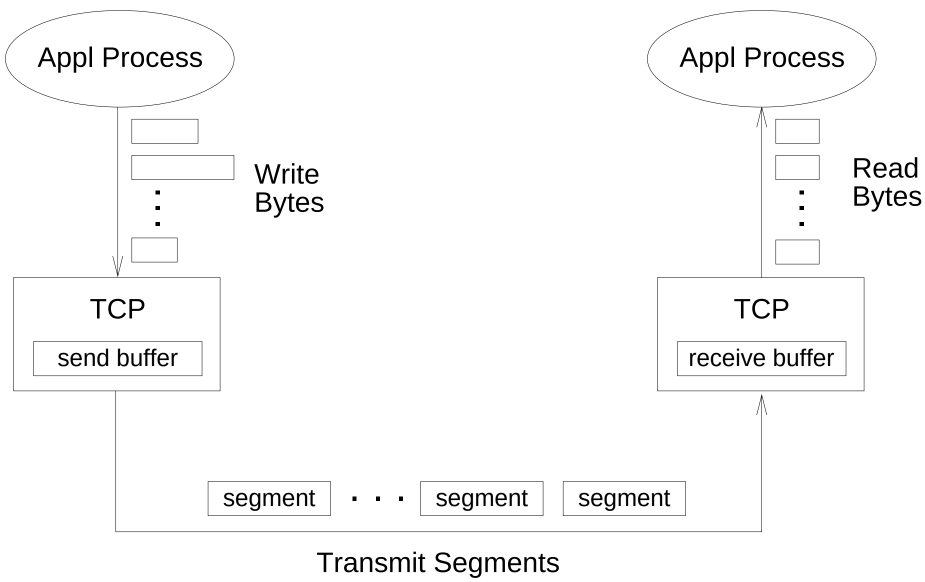
- Unreliable and unordered datagram service
- Adds multiplexing - multiple connections between hosts
- No flow control
- Endpoints identified by ports (16 bits -¿ 64K possible per host)
  - servers have *well-known* ports
  - client and server use these to agree on other port for communication
  - see `/etc/services` on Unix
- Checksum (Optional IPv4, Mandatory IPv6) - same as IP algorithm
  - pseudo header + udp header + udp data
  - pseudo header is IP protocol number, source and destination IP addresses, UDP length field
- Header format

Src Port	Dest Port
Check Sum	Length

Reliable Byte-Stream (TCP)

Overview

- Connection-oriented
- Byte-stream
  - sending process writes some number of bytes
  - TCP breaks into *segments* and sends via IP
  - receiving process reads some number of bytes



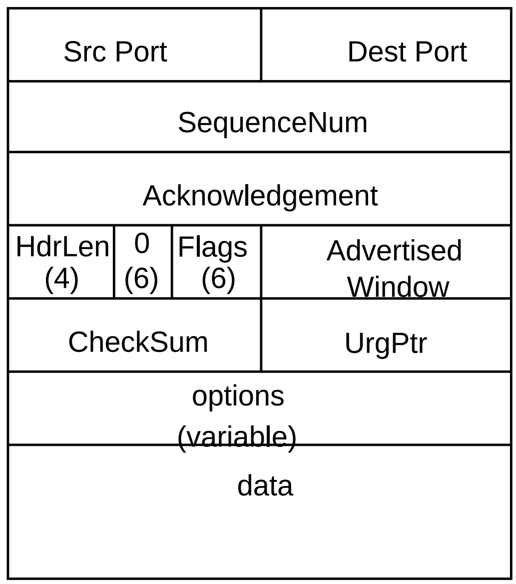
- Full duplex
- Flow control: keep sender from overrunning receiver
- Congestion control: keep sender from overrunning network

End-to-End Issues

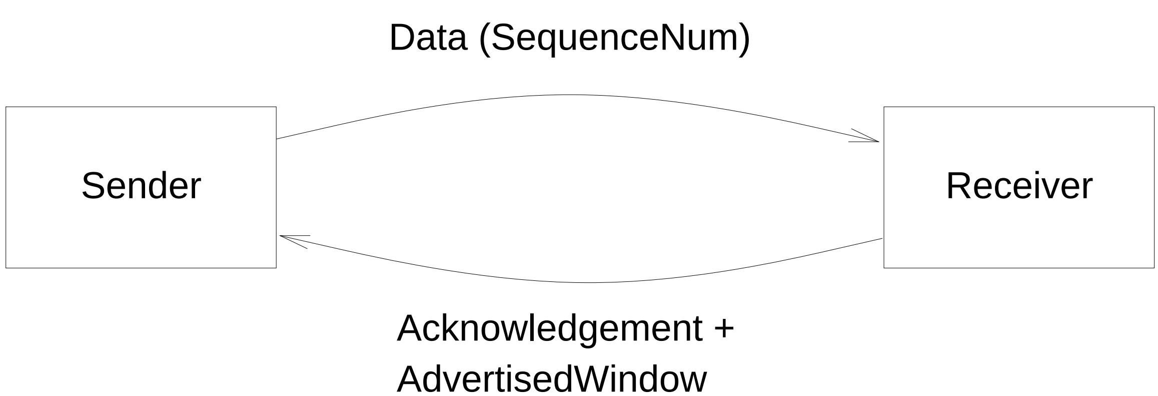
Based on sliding window protocol used at data link level, but the situation is very different.

1. Potentially connects many different hosts
   - need explicit connection establishment and termination
2. Potentially different RTT
   - need adaptive timeout mechanism
3. Potentially long delay in network
   - need to be prepared for arrival of very old packets
   - is limit, discarded after TTL
   - MSL (maximum segment lifetime) - recommended 120 sec
4. Potentially different capacity at destination
   - need to accommodate different amounts of buffering
5. Potentially different network capacity
   - need to be prepared for network congestion

Segment Format



- Each connection identified with 4-tuple:
  - ▶ ⟨`SrcPort, SrcIPAddr, DstPort, DstIPAddr`⟩
- Sliding window + flow control
  - ▶ `Acknowledgment, SequenceNum, AdvertisedWindow`



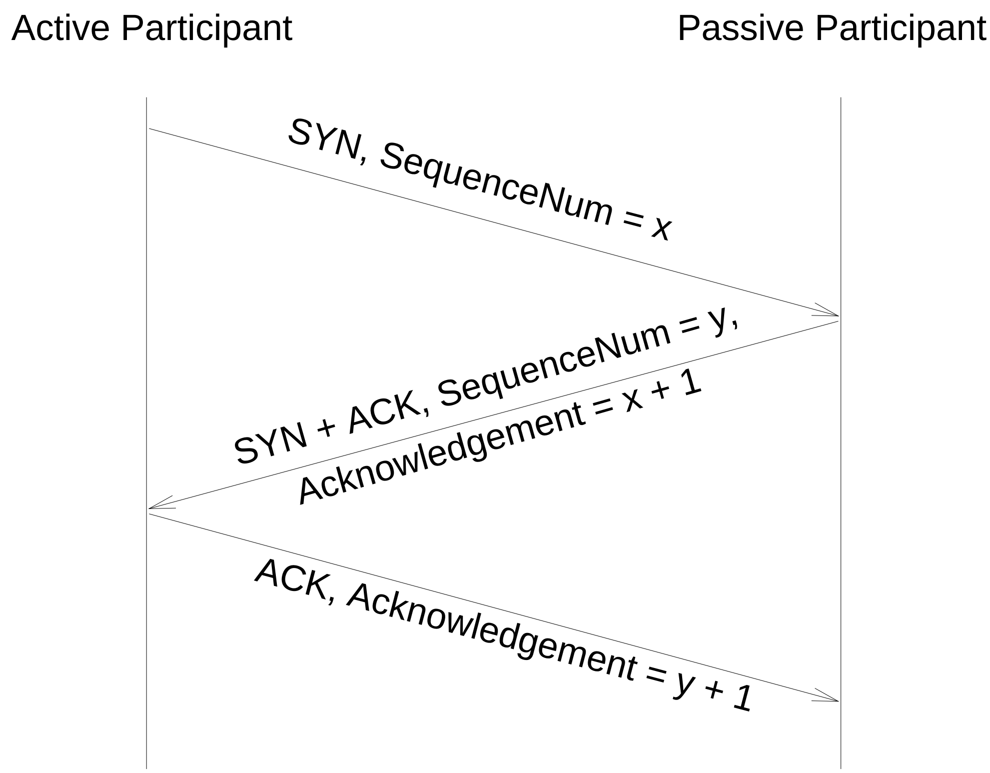
- Flags
  - `SYN`: TCP connection establishing
  - `FIN`: TCP connection terminating
  - `RESET`: Receiver is confused - abort connection
  - `PUSH`: Sender invokes push operation
  - `URG`: segment contains urgent data
  - `ACK`: Acknowledgment
- Checksum
  - pseudo header + tcp header + data

When does TCP send segment?

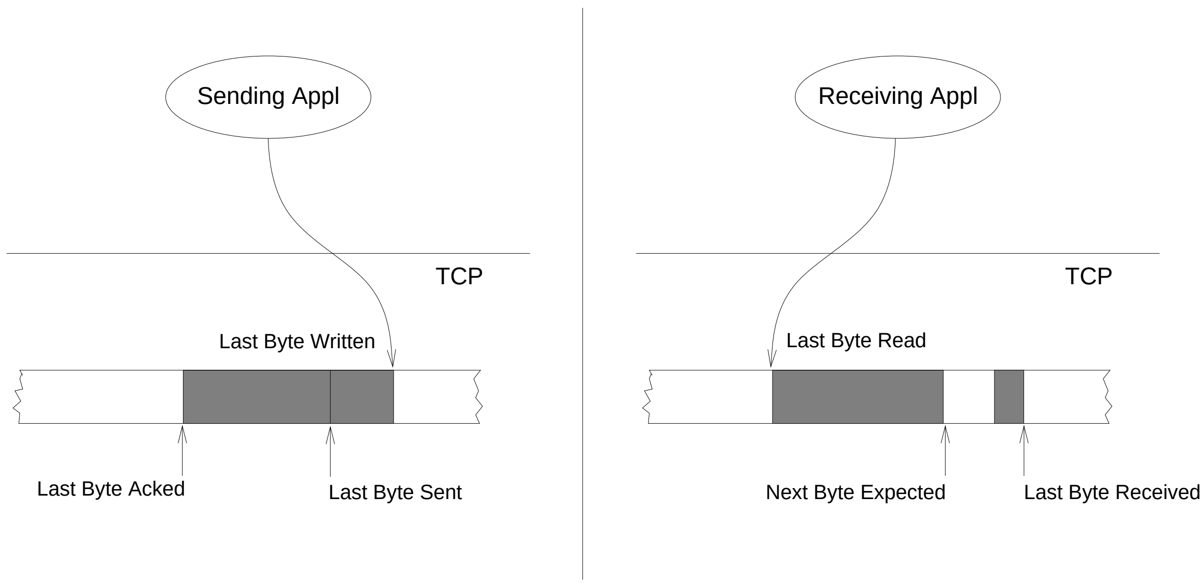
1. After MSS (Maximum Segment Size) bytes are buffered
   - usually largest size that IP will not fragment
   - MSS = MTU - sizeof(TCP + IP headers)
2. if sender flushes buffer with **push** operation
   - Telnet does after each character
3. when timer expires

Connection Establishment and Termination

Three-Way Handshake



Sliding Window Revisited



- Each byte has a sequence number
- ACKs are cumulative
- Sending side
  - `LastByteAcked` $\leq$ `LastByteSent`
  - `LastByteSent` $\leq$ `LastByteWritten`
  - bytes between `LastByteAcked` and `LastByteWritten` must be buffered
- Receiving side
  - LastByteRead < NextByteExpected, Why?
  - `NextByteExpected` $\leq$ `LastByteRcvd + 1`, Why?

Flow Control

- Receiver *advertises* a window size to prevent buffer overflow
- Sender buffer size: `MaxSendBuffer`
- Receive buffer size: `MaxRcvBuffer`
- Receiving side
  - `LastByteRcvd - LastByteRead` $\leq$ `MaxRcvBuffer`
  - `AdvertisedWindow = MaxRcvBuffer - (LastByteRcvd - LastByteRead)`
- Sending side
  - `LastByteSent - LastByteAcked` $\leq$ `AdvertisedWindow`
  - `EffectiveWindow = AdvertisedWindow - (LastByteSent - LastByteAcked)`
  - `LastByteWritten - LastByteAcked` $\leq$ `MaxSendBuffer`
  - TCP blocks sender from sending y bytes if `(LastByteWritten - LastByteAcked) +` $y >$ `MaxSendBuffer`
- Always send ACK in response to an arriving data segment, but not otherwise
- Sender persists in sending 1 byte when `AdvertisedWindow=0`
- Eventually ACK will arrive with new `AdvertisedWindow`

Keeping the Pipe Full

- Wrap Around: 32-bit `SequenceNum` - want no wrap in 120 sec

Bandwidth	Time Until Wrap Around
T1 (1.5Mbps)	6.4 hours
	$2^{32}/(1.544/8)$ bytes $=$6.18 hrs
Ethernet (10Mbps)	57 minutes
T3 (45Mbps)	13 minutes
FDDI (100Mbps)	6 minutes
STS-3 (155Mbps)	4 minutes
STS-12 (622Mbps)	55 seconds
STS-24 (1.2Gbps)	28 seconds

- Bytes in Transit: 16-bit `AdvertisedWindow` allows 64KB of data in pipe
  - Assume RTT$=$ 100 ms typical crosscountry delay in US

Bandwidth	Delay $\times$ Bandwidth Product
T1 (1.5Mbps)	18KB
Ethernet (10Mbps)	122KB
T3 (45Mbps)	549KB
FDDI (100Mbps)	1.2MB
STS-3 (155Mbps)	1.8MB
STS-12 (622Mbps)	7.4MB
STS-24 (1.2Gbps)	14.8MB

Adaptive Retransmission

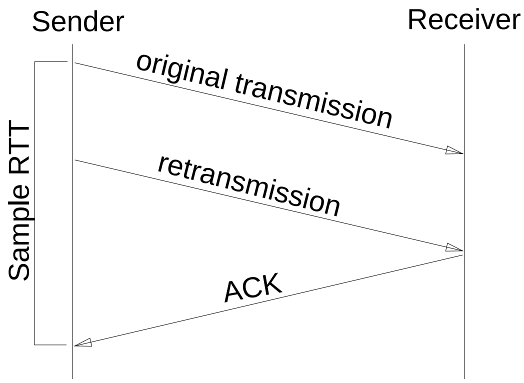
⟹ Original Algorithm

- Measure SampleRTT for each segment/ACK pair
- Compute weighted average of RTT
  - EstimatedRTT = $\alpha\times$ EstimatedRTT + $\beta\times$ SampleRTT
  - where α + β = 1
  - α between 0.8 and 0.9
  - β between 0.1 and 0.2
- Set timeout based on EstimatedRTT
  - TimeOut = 2 $\times$ EstimatedRTT

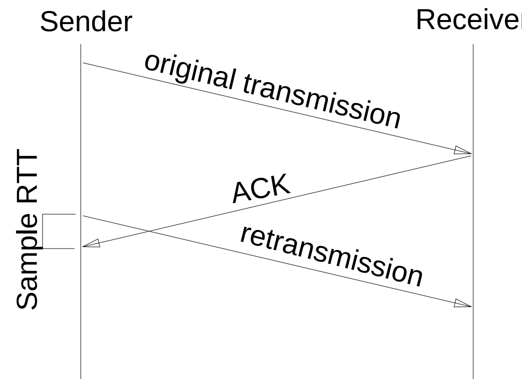
⟹ A flaw

- Does ACK really acknowledges a transmission?
- No, it acknowledges receipt of a segment
- How many retransmissions had taken place before ACK arrived?

⟹ Wrong samples



(a) Sample RTT too long



(b) Sample RTT too short

- in (a) sample should be for the second attempt
- in (b) sample should be for the first attempt

⟹ Karn/Partridge Algorithm

- Do not sample RTT when retransmitting
- Double timeout after each retransmission
  - Similar to backoff algorithm

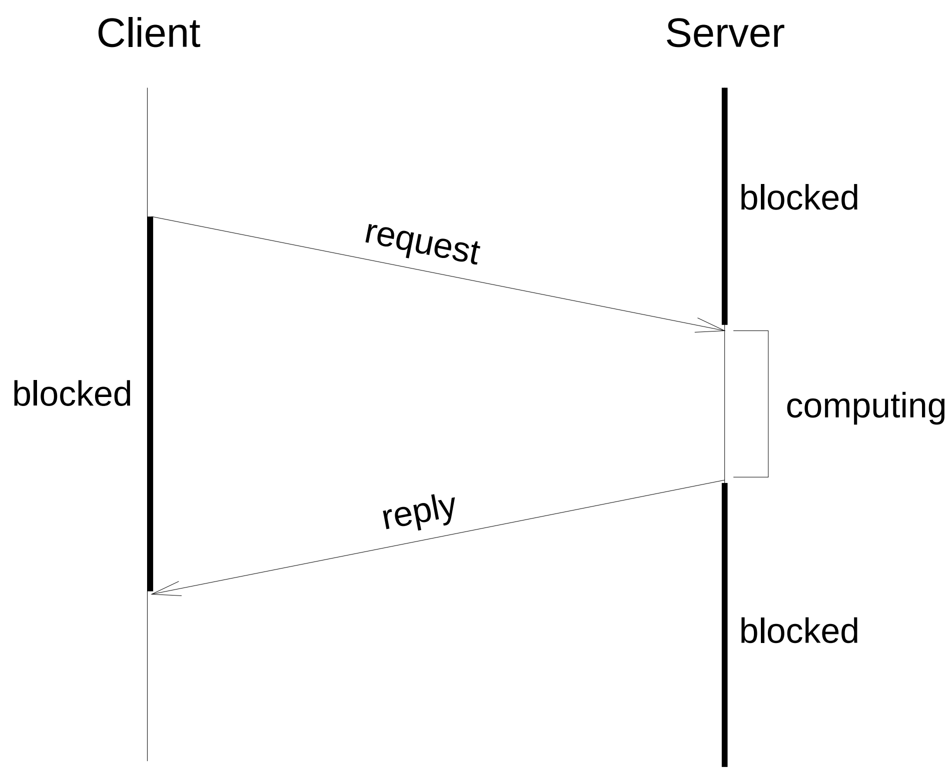
$\Longrightarrow$ Jacobson/Karels Algorithm

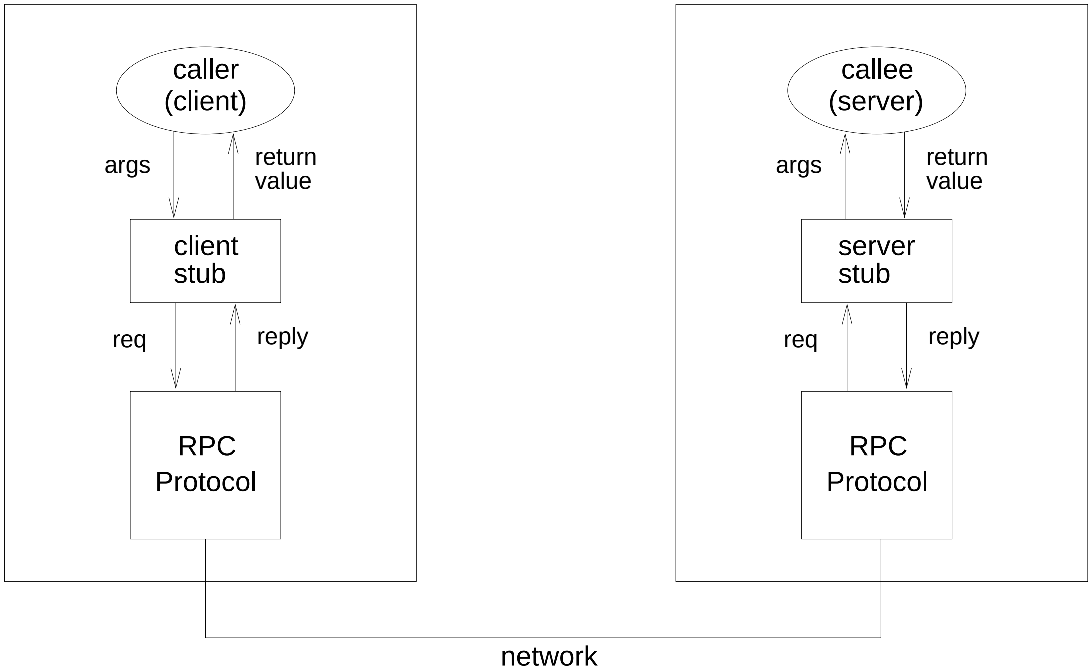
- Karn/Partridge algorithm was introduced when the Internet was not suffering the current congestion
- Consider variance when setting timeout value
- Jacobson/Karels came up with a new calculation for average RTT
  `Difference = SampleRTT - EstimatedRTT`
  `EstimatedRTT = EstimatedRTT +` (δ $\times$ `Difference`)
  `Deviation = Deviation +` δ ($|$`Difference`$|$- `Deviation`)
  - where δ is a fraction between 0 and 1
- `TimeOut =` μ $\times$ `EstimatedRTT +` ϕ $\times$ `Deviation`
  - where $\mu = 1$ and $\phi = 4$

Remote Procedure Call

Overview

- Common pattern of communication used by application programs
- Also called message transaction



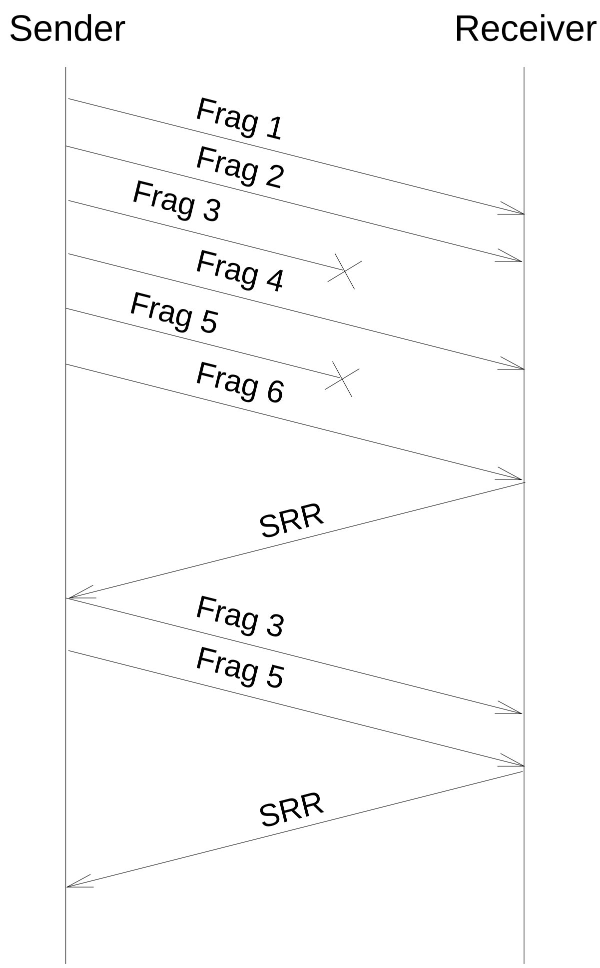


Peterson divides RPC protocol into three basic functions

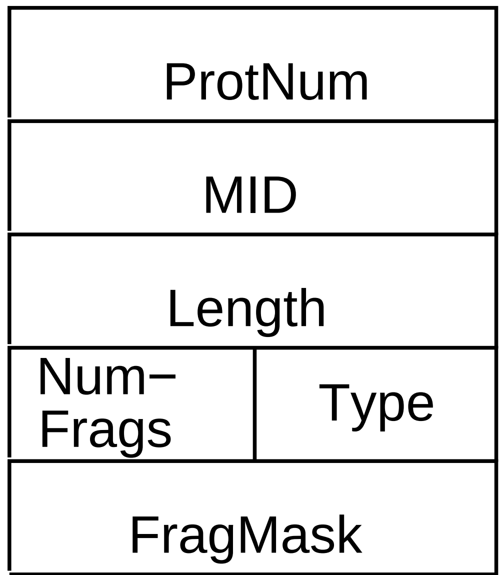
- BLAST: fragments and reassembles large messages
- CHAN: synchronizes request and reply messages
- SELECT: dispatches request messages to the correct process

Bulk Transfer (BLAST)

Unlike AAL and IP in that it tries to recover from lost fragments; persistent, but does not guarantee delivery. Strategy is to use *selective retransmission* (or *partial acknowledgments*).



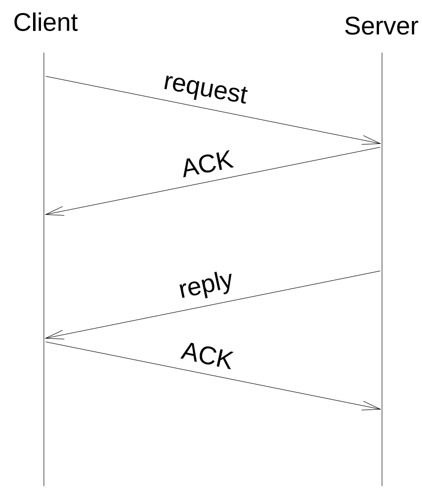
BLAST Header Format



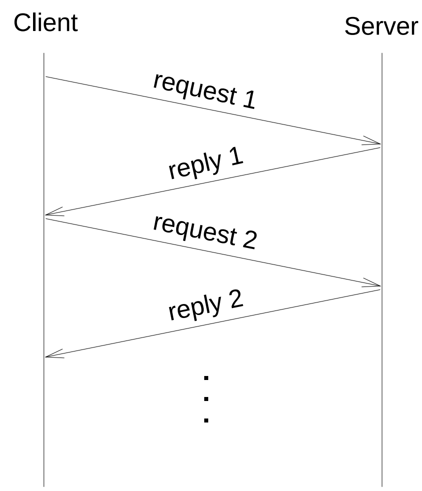
- `MID` must protect against wrap around
- `Type` = `DATA` or `SRR`
- `NumFrags` indicates number of fragments in message
- `FragMask` distinguishes among fragments:
  - if `Type=DATA`, identifies this fragment
  - if `Type=SRR`, identifies missing fragments

Request/Reply (CHAN)

Guarantees message delivery, and synchronizes client with server; i.e., blocks client until reply received. Supports *at-most-once* semantics. Simple case:

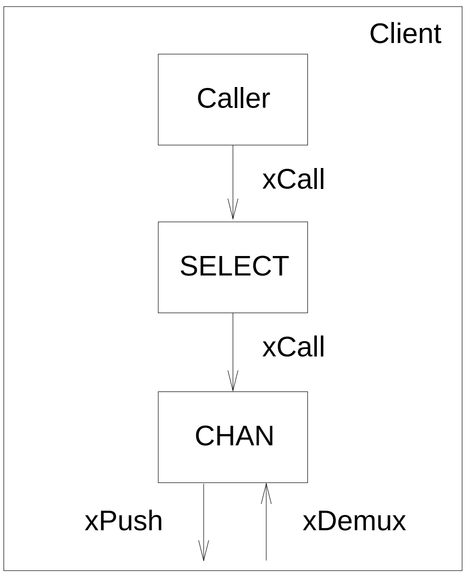


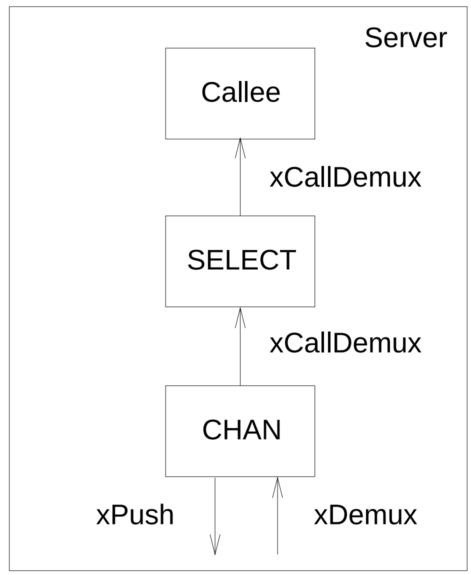
Implicit Acknowledgments:



Dispatcher (SELECT)

Dispatches request messages to the appropriate procedure; fully synchronous counterpart to UDP.





Address Space for Procedures

- Flat: unique id for each possible procedure
- Hierarchical: program + procedure within program

Putting it All Together

Simple RPC Stack